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175th Meeting
Acoustical Society of America

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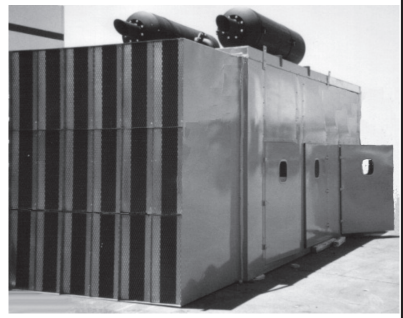

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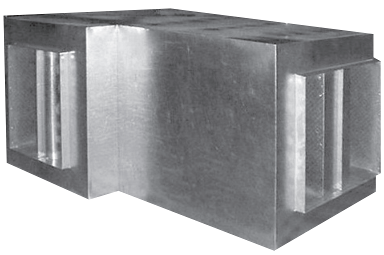

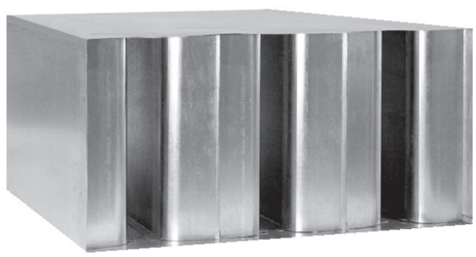

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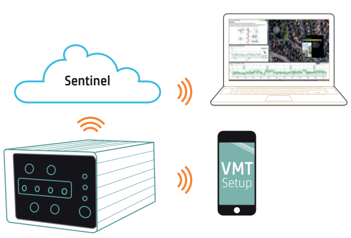

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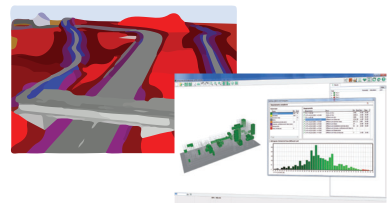

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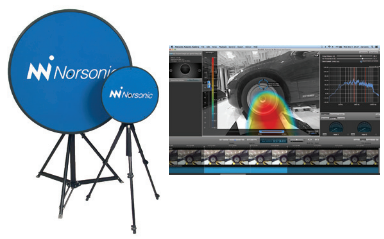

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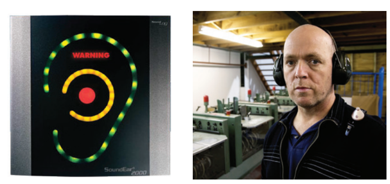

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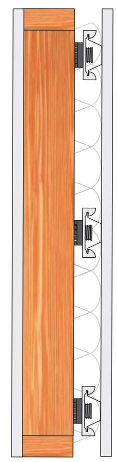

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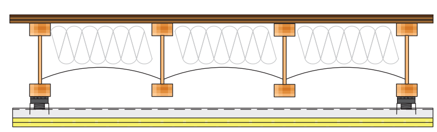

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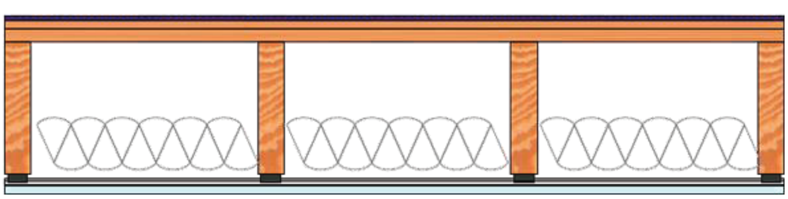

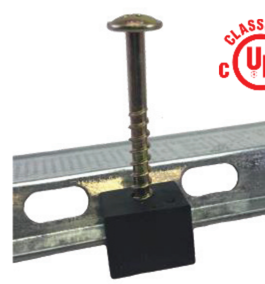

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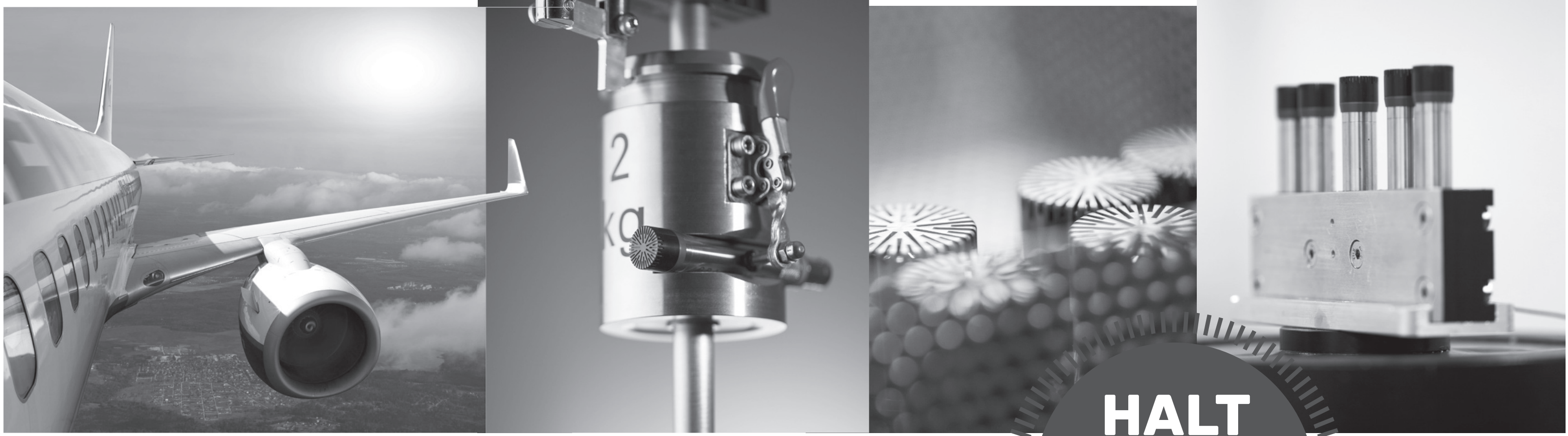

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TECHNICAL PROGRAM SUMMARY

***Indicates Special Session**

MONDAY MORNING

*1aAO	Acoustics in Naturally Constrained Environments: Estuaries, Bays, Inlets, Fjords and Rivers I
*1aNS	Novel Materials for Sound Absorption, Insulation, and Vibration Control
*1aPA	Novel Methods in Computational Acoustics I
*1aPP	Consequences of Asymmetrical Hearing
*1aSP	Signal Processing for Complex Environments I

MONDAY AFTERNOON

*1pAA	Architectural Acoustics Experimental Methods in Laboratories
1pAB	Bat and Biomimetic Sonar/Marine Mammal Hearing and Sound Production
*1pAO	Acoustics in Naturally Constrained Environments: Estuaries, Bays, Inlets, Fjords and Rivers II
*1pBA	Transcranial Focused Ultrasound for Targeted Brain Therapies
1pID	Introductions to Technical Committees
*1pNS	Open Source Audio Processing Platforms
*1pPA	Novel Methods in Computational Acoustics II
*1pPPa	Future Directions for Hearing Aids: Multi-Sensor, User-Informed, and Environment-Aware
1pPPb	Spectral, Pitch, and Physiological Measurements (Poster Session)
1pSA	General Topics in Structural Acoustics and Vibration
*1pSCa	South Asian Languages I
*1pSCb	South Asian Languages II (Poster Session)
*1pSP	Signal Processing for Complex Environments II
1pUW	Instrumentation for Underwater Acoustics

MONDAY EVENING

*1eID	Tutorial Lecture on Hearing Loss and the Future of Auditory Implants

TUESDAY MORNING

*2aAAa	Acoustics of Lobbies, Atria, Stairways, Corridors, Pre-function, etc.
2aAAb	Student Design Competition (Poster Session)
*2aAB	History of Animal Bioacoustics
*2aAOa	Acoustic Seabed Characterization I
2aAOb	Acoustical Oceanography Prize Lecture
*2aBA	Using Acoustic Wave Propagation to Estimate Quantitative Material Properties of Tissue I
*2aEA	Miniature Acoustic Transducers I
*2aNS	Hearing Health Across a Lifespan: Hearing Screening From Cradle to Grave
*2aPA	Infrasound for Global Security I
*2aPP	Phase Locking and Rate Limits in Electric Hearing
*2aSC	Adapting Methods and Models for Vocal Production Across Human and Non-Human Primate Species
*2aSP	Time Reversal Acoustics

TUESDAY AFTERNOON

*2pAA	Interactions Between Acoustics and Architectural Design
*2pAB	Plant Bioacoustics
*2pAO	Acoustic Seabed Characterization II
*2pBA	Using Acoustic Wave Propagation to Estimate Quantitative Material Properties of Tissue II
*2pEA	Miniature Acoustic Transducers II
*2pNS	Effects of Natural Soundscapes on Recreation Areas
*2pPA	Infrasound for Global Security II
*2pPPa	Honoring the Contributions of David Kemp to the Discovery of Otoacoustic Emissons and their Utility for Assessing Hearing Function
2pPPb	Binaural Hearing and Psychoacoustic Methodology (Poster Session)
*2pSA	Improving Education in Structural Acoustics and Vibration
2pSC	Clinical Populations (Poster Session)
*2pSP	Acoustics of Virtual and Augmented Reality

WEDNESDAY MORNING

*3aAA	Auditory Perception in Virtual, Mixed, and Augmented Environments
*3aAO	Ambient Noise Oceanography in Polar Regions: Noise Properties and Parameter Estimation.
*3aBA	Induction Mechanisms for Bubble Nucleation I
3aEAa	General Studies on Structures
3aEAb	General Studies on Transducers
*3aEDa	Developing and/or Using Interactive Simulations for Teaching Acoustics
*3aEDb	Hands-On for Middle School Students
*3aMU	Acoustics of Choirs and Vocal Ensembles
3aPA	General Topics in Physical Acoustics I
*3aPPa	Physiology Meets Perception
3aPPb	Auditory Neuroscience Prize Lecture
*3aSC	Session in Memory of James J. Jenkins
*3aSP	Co-Prime Arrays and Other Sparse Arrays I
*3aUW	Target Scattering in Underwater Acoustics: Imaging, Spectral Domain, and Other Representations I

WEDNESDAY AFTERNOON

*3pAA	AIA CEU Course Presenters Training Session
*3pAB	Lessons on Auditory ‘Perception’ from Exploring Insect Hearing
3pBAa	Biomedical Acoustics Best Student Paper Competition (Poster Session)
*3pBAb	Induction Mechanisms for Bubble Nucleation II
3pID	Hot Topics in Acoustics
*3pPA	Ultrasound and High Frequency Sound in Air in Public and Work Places: Applications, Devices and Effects
*3pPP	Acoustics Outreach to Budding Scientists: Planting Seeds for Future Clinical and Physiological Collaborations ‘18
*3pSA	Model Reduction for Structural Acoustics and Vibration
3pSC	Tools and Technology for Speech Research (Poster Session)
*3pSP	Co-Prime Arrays and Other Sparse Arrays II
*3pUW	Target Scattering in Underwater Acoustics: Imaging, Spectral Domain, and Other Representations II

WEDNESDAY EVENING

*3eED	Listen Up and Get Involved

THURSDAY MORNING

*4aAAa	Acoustics in Classrooms and Other Educational Spaces
4aAAb	Topics in Architectural Acoustics - Measurements, Perceptions, and Metrics
4aAB	Animal sound production and hearing
*4aBA	Acoustic Imaging of Small Vessels and Low Speed Flow
*4aEA	Advanced Transduction Technologies for Sonar Applications
*4aID	Acoustical Standards in Action: Realization, Applications, and Evolution
4aMU	General Topics in Musical Acoustics
*4aNS	Hearing Protection: Impulse Peak Insertion Loss, Specialized Hearing Protection Devices and Fit-Testing I
*4aPA	Sonic Boom I
*4aPP	Honoring Neal Viemeister’s Contributions to Psychoacoustics
*4aSA	Acoustic Metamaterials I
4aSC	Perception and Processing of Voice and Speech (Poster Session)
*4aUW	High Performance Computing Applications to Underwater Acoustics

THURSDAY AFTERNOON

4pBAa	Therapeutic Ultrasound and Bioeffects
4pBAb	Imaging
*4pMU	Sound Effects and Perception
*4pNS	Hearing Protection: Impulse Peak Insertion Loss, Specialized Hearing Protection Devices II
*4pPA	Sonic Boom II
4pPPa	Sound Localization, Spatial Release from Masking, and Binaural Hearing with Devices
4pPPb	Clinical Topics and Devices (Poster Session)
4pPPc	Sensitivity to Spectral and Temporal Information
*4pSA	Acoustic Metamaterials II
4pSC	Non-Native Speech Perception and Production (Poster Session)
4pSP	Topics in Signal Processing in Acoustics
4pUW	Underwater Acoustic Communications, Positioning and Signal Processing

FRIDAY MORNING

5aPA	General Topics in Physical Acoustics II
5aPP	Examining the Auditory Brain
5aSC	Speech Production (Poster Session)
*5aSP	Continuous Active Acoustics
5aUWa	Underwater Soundscape: Measurement and Characterization
5aUWb	Underwater Acoustic Propagation: Models and Experimental Data

SCHEDULE OF STARTING TIMES FOR TECHNICAL SESSIONS AND TECHNICAL COMMITTEE (TC) 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
Greenway A	1aAO 9:30	1pAO 3:00		2aAOa 8:00 2aAOb 11:00	2pAO 1:00	TCAO 7:30	3aAO 8:15	3pSA 1:30					
Greenway B		1pUW 1:45					3aUW 8:00	3pUW 1:00		4aUW 8:30	4pUW 1:00	TCUW 7:30	5aUWa 8:00 5aUWb 9:30
Greenway C		1pSA 1:00			2pSA 1:00	TCSA 7:30		3pID 2:15		4aSA 9:00	4pSA 1:30		
Greenway D				2aEA 9:00	2pEA 1:00	TCEA 4:30	3aEAa 8:15 3aEAb 10:15			4aEA 10:10			
Greenway F/G		1pBA 1:25		2aBA 8:00	2pBA 1:15	TCMU 7:30	3aBA 9:00	3pBAb 1:40	TCBA 7:30	4aBA 8:00	4pBAa 1:30 4pBAb 3:30		
Greenway H/I	1aSP 9:00	1pSP 1:00		2aSP 8:20	2pSP 1:00		3aSP 8:00	3pSP 1:00	TCSP 7:30	4aID 8:00	4pSP 1:00		5aSP 8:00
Greenway J	1aPA 8:40	1pPA 1:00		2aPA 9:40	2pPA 1:00	TCPA 7:30	3aPA 9:00	3pPA 1:00		4aPA 8:40	4pPA 1:40		5aPA 8:00
Lakeshore A		1pID 1:00					3aMU 9:00	3pAA 1:00		4aMU 8:30	4pMU 1:00		
Lakeshore B		1pAB 1:20		2aAB 8:00	2pAB 1:20	TCAB 7:30		3pAB 1:10		4aAB 8:00			
Mirage							3aEDb 9:30		3eED 5:00				
Nicolette A		1pPPb 1:00 1pSCb 2:15		2aAAb 9:00	2pPPb 1:00 2pSC 1:00			3pBAa 1:00 3pSC 1:00		4aSC 8:00	4pPPb 1:00 4pSC 1:00		5aSC 8:00
Nicolette C		1pAA 1:00		2aAAa 9:00	2pAA 1:00		3aAA 8:50			4aAAa 7:55 4aAAb 10:10			
Nicolette D1		1pSCa 1:00		2aSC 8:40			3aSC 8:40						
Nicolette D2	1aPP 8:00	1pPPa 1:00	1eID 7:00	2aPP 8:00	2pPPa 1:00	TCPP 7:30	3aPPa 7:55 3aPPb 11:00	3pPP 1:10		4aPP 8:00	4pPPa 1:00 4pPPc 3:45	TCSC 7:30	5aPP 8:00
Nicolette D3	1aNS 8:30	1pNS 1:00		2aNS 8:30	2pNS 1:00	TCAA 7:30	3aEDa 8:00			4aNS 8:00	4pNS 1:30	TCNS 7:30	

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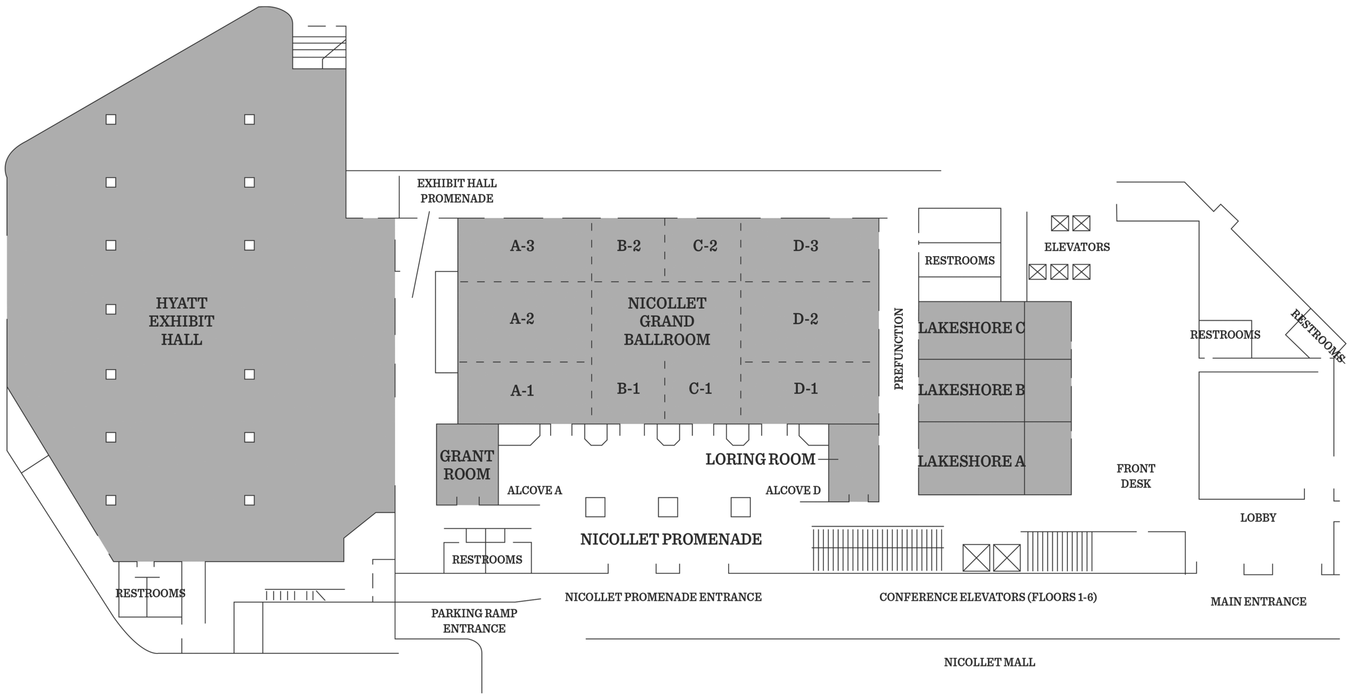

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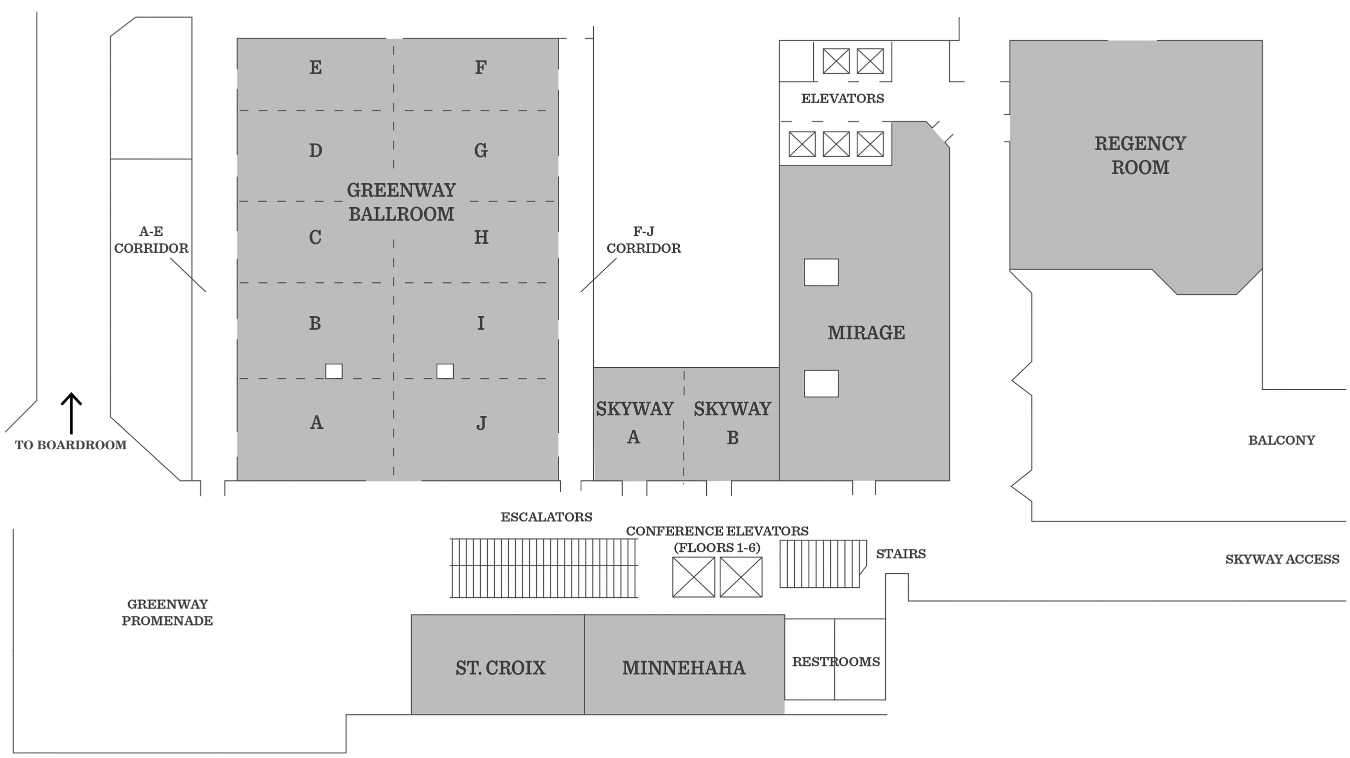

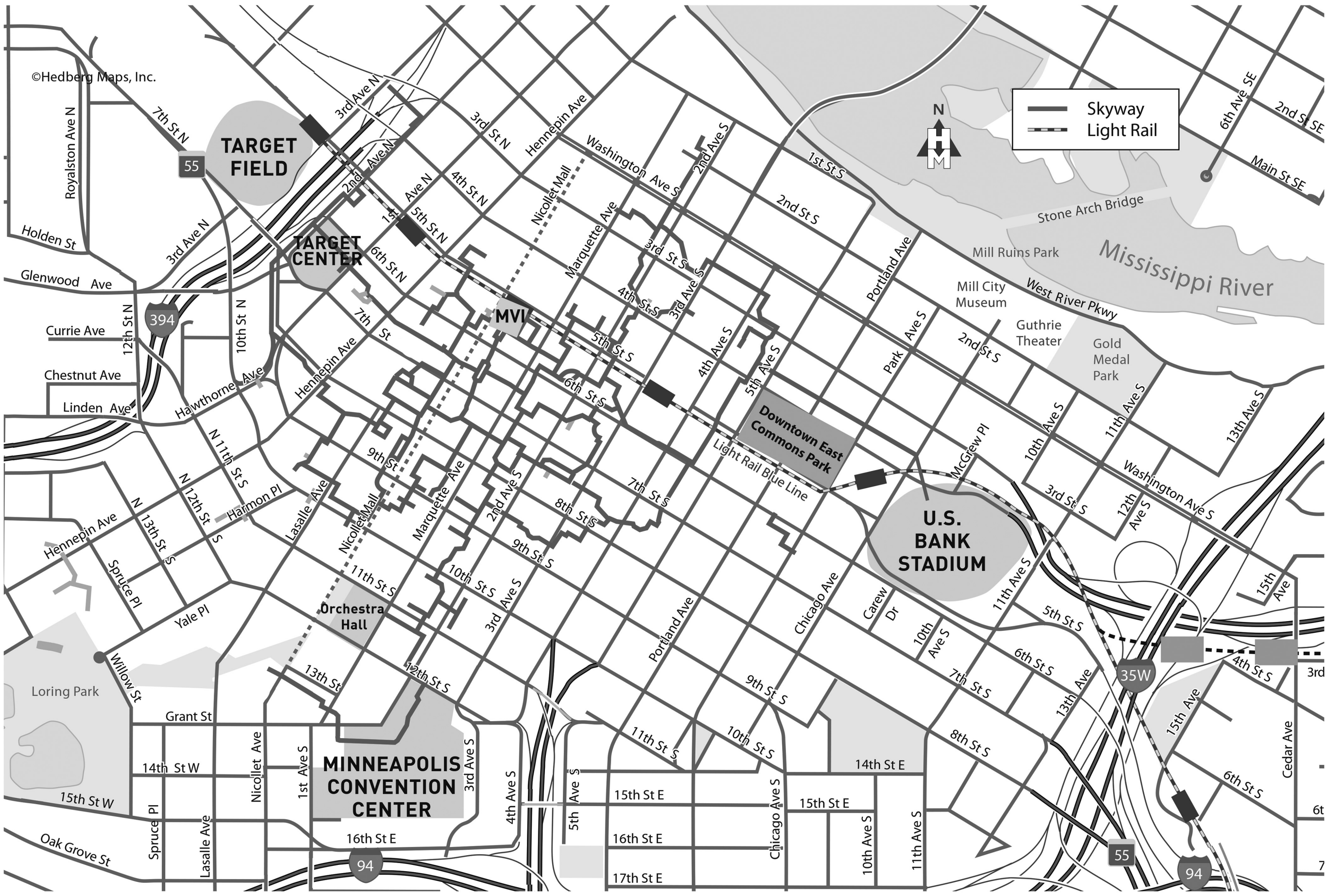
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Light Rail
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TARGET FIELD
TARGET CENTER
MVI
Downtown East Commons Park
U.S. BANK STADIUM
Orchestra Hall
MINNEAPOLIS CONVENTION CENTER
Mississippi River
Stone Arch Bridge
Mill Ruins Park
Mill City Museum
Guthrie Theater
Gold Medal Park
Loring Park
Light Rail Blue Line
West River Pkwy
Royalston Ave N
7th St N
3rd Ave N
Holden St
Glenwood Ave
Currie Ave
Chestnut Ave
Linden Ave
12th St N
10th St N
Hawthorne Ave
Hennepin Ave
Harmon Pl
Lasalle Ave
Nicollet Mall
Marquette Ave
2nd Ave S
3rd Ave S
4th Ave S
5th Ave S
Portland Ave
Park Ave S
Chicago Ave
Carew Dr
10th Ave S
11th Ave S
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13th Ave S
15th Ave
Cedar Ave
McGrew Pl
6th Ave SE
2nd St SE
Main St SE
Washington Ave S
1st St S
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3rd St S
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Grant St
14th St W
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Oak Grove St
Nicollet Ave
1st Ave S
14th St E
15th St E
16th St E
17th St E
Chicago Ave S
55
394
94
35W

TECHNICAL PROGRAM CALENDAR
175th Meeting of the Acoustical Society of America
7–11 May 2018

MONDAY MORNING

9:30 1aAO **Acoustical Oceanography and Underwater Acoustics:** Acoustics in Naturally Constrained Environments: Estuaries, Bays, Inlets, Fjords and Rivers I. Greenway A

8:30 1aNS **Noise:** Novel Materials for Sound Absorption, Insulation, and Vibration Control. Nicollet D3

8:40 1aPA **Physical Acoustics, Structural Acoustics and Vibration, Architectural Acoustics, and Underwater Acoustics:** Novel Methods in Computational Acoustics I. Greenway J

8:00 1aPP **Psychological and Physiological Acoustics:** Consequences of Asymmetrical Hearing. Nicollet D2

9:00 1aSP **Signal Processing in Acoustics, Underwater Acoustics, and Architectural Acoustics:** Signal Processing for Complex Environments I. Greenway H/I

MONDAY AFTERNOON

1:00 1pAA **Architectural Acoustics and Noise:** Architectural Acoustics Experimental Methods in Laboratories. Nicollet C

1:20 1pAB **Animal Bioacoustics:** Bat and Biomimetic Sonar/Marine Mammal Hearing and Sound Production. Lakeshore B

3:00 1pAO **Acoustical Oceanography and Underwater Acoustics:** Acoustics in Naturally Constrained Environments: Estuaries, Bays, Inlets, Fjords and Rivers II. Greenway A

1:25 1pBA **Biomedical Acoustics:** Transcranial Focused Ultrasound for Targeted Brain Therapies. Greenway F/G

1:00 1pID **Interdisciplinary:** Introductions to Technical Committees. Lakeshore A

1:00 1pNS **Noise:** Open Source Audio Processing Platforms. Nicollet D3

1:00 1pPA **Physical Acoustics, Structural Acoustics and Vibration, Architectural Acoustics, and Underwater Acoustics:** Novel Methods in Computational Acoustics II. Greenway J

1:00 1pPPa **Psychological and Physiological Acoustics, Signal Processing in Acoustics, and Underwater Acoustics:** Future Directions for Hearing Aids: Multi-Sensor, User-Informed, and Environment-Aware. Nicollet D2

1:00 1pPPb **Psychological and Physiological Acoustics:** Spectral, Pitch, and Physiological Measurements (Poster Session). Nicollet A

1:00 1pSA **Structural Acoustics and Vibration:** General Topics in Structural Acoustics and Vibration. Greenway C

1:00 1pSCa **Speech Communication:** South Asian Languages I. Nicollet D1

2:15 1pSCb **Speech Communication:** South Asian Languages II (Poster Session). Nicollet A

1:00 1pSP **Signal Processing in Acoustics, Underwater Acoustics, and Architectural Acoustics:** Signal Processing for Complex Environments II. Greenway H/I

1:45 1pUW **Underwater Acoustics:** Instrumentation for Underwater Acoustics. Greenway B

MONDAY EVENING

7:00 1eID **Interdisciplinary:** Tutorial Lecture on Hearing Loss and the Future of Auditory Implants. Nicollet D2

TUESDAY MORNING

9:00 2aAAa **Architectural Acoustics and ASA Committee on Standards:** Acoustics of Lobbies, Atria, Stairways, Corridors, Pre-function, etc.. Nicollet C

9:00 2aAAb **Architectural Acoustics:** Student Design Competition (Poster Session). Nicollet A

8:00 2aAB **Animal Bioacoustics and ASA Committee on Standards:** History of Animal Bioacoustics. Lakeshore B

8:00 2aAOa **Acoustical Oceanography:** Acoustic Seabed Characterization I. Greenway A

11:00 2aAOb **Acoustical Oceanography:** Acoustical Oceanography Prize Lecture. Greenway A

8:00 2aBA **Biomedical Acoustics:** Using Acoustic Wave Propagation to Estimate Quantitative Material Properties of Tissue I. Greenway F/G

9:00 2aEA **Engineering Acoustics and Physical Acoustics:** Miniature Acoustic Transducers I. Greenway D

8:30 2aNS **Noise, Psychological and Physiological Acoustics, and Speech Communication:** Hearing Health Across a Lifespan: Hearing Screening From Cradle to Grave. Nicollet D3

9:40 2aPA **Physical Acoustics:** Infrasound for Global Security I. Greenway J

8:00 2aPP **Psychological and Physiological Acoustics, Speech Communication, and Signal Processing in Acoustics:** Phase Locking and Rate Limits in Electric Hearing. Nicollet D2

8:40 2aSC **Speech Communication, Animal Bioacoustics, and Psychological and Physiological Acoustics:** Adapting Methods and Models for Vocal Production Across Human and Non-Human Primate Species. Nicollet D1

8:20 2aSP **Signal Processing in Acoustics, Underwater Acoustics, Physical Acoustics, and Noise:** Time Reversal Acoustics. Greenway H/I

TUESDAY AFTERNOON

1:00 2pAA **Architectural Acoustics:** Interactions Between Acoustics and Architectural Design. Nicollet C

1:20 2pAB **Animal Bioacoustics:** Plant Bioacoustics. Lakeshore B

1:00 2pAO **Acoustical Oceanography:** Acoustic Seabed Characterization II. Greenway A

1:15 2pBA **Biomedical Acoustics:** Using Acoustic Wave Propagation to Estimate Quantitative Material Properties of Tissue II. Greenway F/G

1:00 2pEA **Engineering Acoustics and Physical Acoustics:** Miniature Acoustic Transducers II. Greenway D

1:00 2pNS **Noise and ASA Committee on Standards:** Effects of Natural Soundscapes on Recreation Areas. Nicollet D3

1:00 2pPA **Physical Acoustics:** Infrasound for Global Security II. Greenway J

1:00 2pPPa **Psychological and Physiological Acoustic, Animal Bioacoustics, and Noise:** Honoring the Contributions of David Kemp to the Discovery of Otoacoustic Emissions and their Utility for Assessing Hearing Function. Nicollet D2

1:00 2pPPb **Psychological and Physiological Acoustics:** Binaural Hearing and Psychoacoustic Methodology (Poster Session). Nicollet A

1:00 2pSA **Structural Acoustics and Vibration and Education in Acoustics:** Improving Education in Structural Acoustics and Vibration. Greenway C

1:00 2pSC **Speech Communication:** Clinical Populations (Poster Session). Nicollet A

1:00 2pSP **Signal Processing in Acoustics, Psychological and Physiological Acoustics, Architectural Acoustics, and Noise:** Acoustics of Virtual and Augmented Reality. Greenway H/I

WEDNESDAY MORNING

8:50 3aAA **Architectural Acoustics, Psychological and Physiological Acoustics, and Speech Communication:** Auditory Perception in Virtual, Mixed, and Augmented Environments. Nicollet C

8:15 3aAO **Acoustical Oceanography, Animal Bioacoustics, and Underwater Acoustics:** Ambient Noise Oceanography in Polar Regions: Noise Properties and Parameter Estimation. Greenway A

9:00 3aBA **Biomedical Acoustics and Physical Acoustics:** Induction Mechanisms for Bubble Nucleation I. Greenway F/G

8:15 3aEAa **Engineering Acoustics:** General Studies on Structures. Greenway D

10:15 3aEAb **Engineering Acoustics:** General Studies on Transducers. Greenway D

8:00 3aEDa **Education in Acoustics:** Developing and/or Using Interactive Simulations for Teaching Acoustics. Nicollet D3

9:30 3aEDb **Education in Acoustics:** Hands-On for Middle and High-School Students. Mirage

9:00 3aMU **Musical Acoustics:** Acoustics of Choirs and Vocal Ensembles. Lakeshore A

9:00 3aPA **Physical Acoustics:** General Topics in Physical Acoustics I. Greenway J

7:55 3aPPa **Psychological and Physiological Acoustics:** Physiology Meets Perception. Nicollet D2

11:00 3aPPb **Psychological and Physiological Acoustics:** Auditory Neuroscience Prize Lecture. Nicollet D2

8:40 3aSC **Speech Communication:** Session in Memory of James J. Jenkins. Nicollet D1

9:00 3aSP **Signal Processing in Acoustics and Underwater Acoustics:** Co-Prime Arrays and Other Sparse Arrays I. Greenway H/I

8:00 3aUW **Underwater Acoustics:** Target Scattering in Underwater Acoustics: Imaging, Spectral Domain, and Other Representations I. Greenway B

WEDNESDAY AFTERNOON

1:00 3pAA **Architectural Acoustics:** AIA CEU Course Presenters Training Session. Lakeshore A

1:10 3pAB **Animal Bioacoustics:** Lessons on Auditory ‘Perception’ from Exploring Insect Hearing. Lakeshore B

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

1:40 3pBAb **Biomedical Acoustics and Physical Acoustics:** Induction Mechanisms for Bubble Nucleation II. Greenway F/G

2:15 3pID **Interdisciplinary:** Hot Topics in Acoustics. Greenway C/D

1:00 3pPA **Physical Acoustics:** Ultrasound and High Frequency Sound in Air in Public and Work Places: Applications, Devices and Effects. Greenway J

1:10 3pPP **Psychological and Physiological Acoustics and Speech Communication:** Acoustics Outreach to Budding Scientists: Planting Seeds for Future Clinical and Physiological Collaborations '18. Nicollet D2

1:30 3pSA **Structural Acoustics and Vibration, Signal Processing in Acoustics, and Noise:** Model Reduction for Structural Acoustics and Vibration. Greenway A

1:00 3pSC **Speech Communication:** Tools and Technology for Speech Research (Poster Session). Nicollet A

1:00 3pSP **Signal Processing in Acoustics and Underwater Acoustics:** Co-Prime Arrays and Other Sparse Arrays II. Greenway H/I

1:00 3pUW **Underwater Acoustics:** Target Scattering in Underwater Acoustics: Imaging, Spectral Domain, and Other Representations II. Greenway B

WEDNESDAY EVENING

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

THURSDAY MORNING

7:55 4aAAa **Architectural Acoustics, Noise, ASA Committee on Standards, and Speech Communication:** Acoustics in Classrooms and Other Educational Spaces. Nicollet C

10:10 4aAAb **Architectural Acoustics:** Topics in Architectural Acoustics - Measurements, Perceptions, and Metrics. Nicollet C

8:00 4aAB **Animal Bioacoustics:** Animal sound production and hearing. Lakeshore B

8:00 4aBA **Biomedical Acoustics:** Acoustic Imaging of Small Vessels and Low Speed Flow. Greenway F/G

10:10 4aEA **Engineering Acoustics, Signal Processing in Acoustics, and Underwater Acoustics:** Advanced Transduction Technologies for Sonar Applications. Greenway D

8:00 4aID **Interdisciplinary, ASA Committee on Standards, Noise, Structural Acoustics and Vibration, and Architectural Acoustics:** Acoustical Standards in Action: Realization, Applications, and Evolution. Greenway H/I

8:30 4aMU **Musical Acoustics:** General Topics in Musical Acoustics. Lakeshore A

8:00 4aNS **Noise, Psychological and Physiological Acoustics, and ASA Committee on Standards:** Hearing Protection: Impulse Peak Insertion Loss, Specialized Hearing Protection Devices and Fit-Testing I. Nicollet D3

8:40 4aPA **Physical Acoustics, Noise, and ASA Committee on Standards:** Sonic Boom I. Greenway J

8:00 4aPP **Psychological and Physiological Acoustics:** Honoring Neal Viemeister's Contributions to Psychoacoustics. Nicollet D2

9:00 4aSA **Structural Acoustics and Vibration, Physical Acoustics, and Engineering Acoustics:** Acoustic Metamaterials I. Greenway C

8:00 4aSC **Speech Communication:** Perception and Processing of Voice and Speech (Poster Session). Nicollet A

8:30 4aUW **Underwater Acoustics, Signal Processing in Acoustics, and Physical Acoustics:** High Performance Computing Applications to Underwater Acoustics. Greenway B

THURSDAY AFTERNOON

1:30 4pBAa **Biomedical Acoustics:** Therapeutic Ultrasound and Bioeffects. Greenway F/G

3:30 4pBAb **Biomedical Acoustics:** Imaging. Greenway F/G

1:00 4pMU **Musical Acoustics, Psychological and Physiological Acoustics, and Signal Processing in Acoustics:** Sound Effects and Perception. Lakeshore A

1:30 4pNS **Noise, Psychological and Physiological Acoustics, and ASA Committee on Standards:** Hearing Protection: Impulse Peak Insertion Loss, Specialized Hearing Protection Devices II. Nicollet D3

1:40 4pPA **Physical Acoustics, Noise, and ASA Committee on Standards:** Sonic Boom II. Greenway J

1:00 4pPPa **Psychological and Physiological Acoustics:** Sound Localization, Spatial Release from Masking, and Binaural Hearing with Devices. Nicollet D2

1:00 4pPPb **Psychological and Physiological Acoustics:** Clinical Topics and Devices (Poster Session). Nicollet A

3:45 4pPPc **Psychological and Physiological Acoustics:** Sensitivity to Spectral and Temporal Information. Nicollet D2

1:30 4pSA **Structural Acoustics and Vibration, Physical Acoustics, and Engineering Acoustics:** Acoustic Metamaterials II. Greenway C

1:00 4pSC **Speech Communication:** Non-Native Speech Perception and Production (Poster Session). Nicollet A

1:00 4pSP **Signal Processing in Acoustics:** Topics in Signal Processing in Acoustics. Greenway H/I

1:00 4pUW **Underwater Acoustics:** Underwater Acoustic Communications, Positioning and Signal Processing. Greenway B

FRIDAY MORNING

8:00 5aPA **Physical Acoustics:** General Topics in Physical Acoustics II. Greenway J

8:00 5aPP **Psychological and Physiological Acoustics:** Examining the Auditory Brain. Nicollet D2

8:00 5aSC **Speech Communication:** Speech Production (Poster Session). Nicollet A

8:00 5aSP **Signal Processing in Acoustics and Underwater Acoustics:** Continuous Active Acoustics. Greenway H/I

8:00 5aUWa **Underwater Acoustics:** Underwater Soundscape: Measurement and Characterization. Greenway B

9:30 5aUWb **Underwater Acoustics:** Underwater Acoustic Propagation: models and experimental data. Greenway B

SCHEDULE OF COMMITTEE MEETINGS AND OTHER EVENTS

ASA COUNCIL AND ADMINISTRATIVE COMMITTEES

Mon, 7 May, 7:30 a.m.	Executive Council	Minnehaha
Mon, 7 May, 3:30 p.m.	Technical Council	Minnehaha
Mon, 7 May, 6:30 p.m.	Technical Council Dinner	Regency
Tue, 8 May, 7:00 a.m.	ASA Books	Conference Rm, 2nd fl
Tue, 8 May, 7:30 a.m.	Panel on Public Policy	Skyway A
Tue, 8 May, 11:45 a.m.	Editorial Board	Regency
Tue, 8 May, 12:00 noon	Student Council	St. Croix
Tue, 8 May, 12:30 p.m.	Prizes & Special Fellowships	Conference Rm, 2nd fl
Tue, 8 May, 1:30 p.m.	Meetings	Skyway A/B
Tue, 8 May, 4:00 p.m.	Newman Fund Advisory	Conference Rm, 2nd fl
Tue, 8 May, 5:00 p.m.	Women in Acoustics	St. Croix
Tue, 8 May, 5:00 p.m.	International Liaison	Skyway A/B
Wed, 9 May, 6:45 a.m.	International Research & Education	Skyway A
Wed, 9 May, 7:00 a.m.	Archives & History	Skyway B
Wed, 9 May, 7:00 a.m.	College of Fellows	Conference Rm, 2nd fl
Wed, 9 May, 7:00 a.m.	Publication Policy	Lakeshore B
Wed, 9 May, 7:00 a.m.	Regional and Student Chapters	Minnehaha
Wed, 9 May, 7:30 a.m.	Finance	St. Croix
Wed, 9 May, 11:00 a.m.	Medals and Awards	Minnehaha
Wed, 9 May, 11:30 a.m.	Public Relations	St. Croix
Wed, 9 May, 12:00 noon	Membership	Skyway B
Wed, 9 May, 12:00 p.m.	Audit	Conference Rm, 2nd fl
Wed, 9 May, 1:30 p.m.	AS Foundation Board	St. Croix
Wed, 9 May, 5:00 p.m.	Education in Acoustics	Greenway A
Wed, 9 May, 5:00 p.m.	Acoustics Today Editorial Board	Conference Rm, 2nd fl
Wed, 9 May, 5:00 p.m.	TCAA Speech Privacy	Minnehaha
Thu, 10 May, 7:00 a.m.	Investments	Conference Rm, 2nd fl
Thu, 10 May, 7:30 a.m.	Tutorials, Short Courses, Hot Topics	Skyway A
Thu, 10 May, 2:00 p.m.	Strategic Plan Champions	Minnehaha
Thu, 10 May, 4:30 p.m.	Financial Affairs	Skyway B
Thu, 10 May, 4:30 p.m.	Member Engagement and Diversity	Conference Rm, 2nd fl
Thu, 10 May, 4:30 p.m.	Outreach	St. Croix
Thu, 10 May, 4:30 p.m.	Publications and Standards	Skyway A
Fri, 11 May, 7:00 a.m.	Technical Council	Minnehaha
Fri, 11 May, 11:00 a.m.	Executive Council	Minnehaha

TECHNICAL COMMITTEEE OPEN MEETINGS

Tue, 8 May, 4:30 p.m.	Engineering Acoustics	Greenway D
Tue, 8 May, 7:30 p.m.	Acoustical Oceanography	Greenway A
Tue, 8 May, 7:30 p.m.	Animal Bioacoustics	Lakeshore B
Tue, 8 May, 7:30 p.m.	Architectural Acoustics	Nicollet D3
Tue, 8 May, 7:30 p.m.	Musical Acoustics	Greenway F/G
Tue, 8 May, 7:30 p.m.	Physical Acoustics	Greenway J
Tue, 8 May, 7:30 p.m.	Psychological and Physiological Acoustics	Nicollet D2
Tue, 8 May, 7:30 p.m.	Structural Acoustics and Vibration	Greenway C
Wed, 9 May, 7:30 p.m.	Biomedical Acoustics	Greenway F/G
Wed, 9 May, 7:30 p.m.	Signal Processing in Acoustics	Greenway H/I
Thu, 10 May, 7:30 p.m.	Noise	Nicollet D3
Thu, 10 May, 7:30 p.m.	Speech Communication	Nicollet D2
Thu, 10 May, 7:30 p.m.	Underwater Acoustics	Greenway B

STANDARDS COMMITTEES AND WORKING GROUPS

Mon, 7 May, 5:00 p.m.	S2-Mechanical Vibration and Shock	Skyway B
Mon, 7 May, 7:00 p.m.	ASACOS Steering	Skyway B
Tue, 8 May, 7:30 a.m.	ASACOS	Minnehaha
Tue, 8 May, 9:15 a.m.	ASACOS Plenary	Minnehaha
Tue, 8 May, 11:00 a.m.	S12-Noise	Minnehaha
Tue, 8 May, 2:00 p.m.	S3/SC1-Animal Bioacoustics	Minnehaha
Tue, 8 May, 3:15 p.m.	S3-Bioacoustics	Minnehaha
Tue, 8 May, 4:45 p.m.	S1-Acoustics	Minnehaha
Wed, 9 May, 8:30 a.m.	S12/WG11-Hearing Protectors	Skyway A
Wed, 9 May, 1:00 p.m.	S3/WG47	Skyway A

MEEETING SERVICES, SPECIAL EVENTS, SOCIAL EVENTS

Sun, 6 May 9:00 a.m. - 5:00 p.m.	International Quiet Ocean Experiment Forum	Minnehaha
Sun, 6 May 1:00 p.m. - 5:00 p.m. Mon, 7 May 8:00 a.m. - 12:30 p.m.	Short Course	Mirage
Mon-Thu, 7-10 May 7:30 a.m. - 5:00 p.m. Fri, 11 May 7:30 a.m. - 12:00 noon	Registration	Exhibit Hall
Mon-Thu, 7-10 May, 7:00 a.m. - 5:00 p.m. Fri, 11 May, 7:00 a.m. - 12:00 noon	E-mail	Nicollet Promenade
Mon-Thu, 7-10 May, 7:00 a.m. - 5:00 p.m. Fri, 11 May, 7:00 a.m. - 12:00 noon	Internet Zone	Nicollet Promenade
Mon-Thu, 7-10 May, 7:00 a.m. - 5:30 p.m. Fri, 11 May, 7:00 a.m. - 12:00 noon	A/V Preview	Greenway E
Mon-Thu, 7-11 May 8:00 a.m. - 10:00 a.m.	Accompanying Persons	Regency
Mon-Thu, 7-10 May 9:00 a.m. - 5:00 p.m.	Gallery of Acoustics	Nicollet Promenade
Mon, Thu, Fri, 7, 10, 11 May	Morning Coffee Break	Nicollet Foyer
Tue-Wed, 8-9 May, 9:30 a.m. - 11:00 a.m.		Exhibit Hall
Tue, 8 May 2:30 p.m. - 3:30 p.m.	Afternoon Coffee Break	Exhibit Hall
Mon, 7 May 5:30 p.m. - 7:00 p.m.	Exhibit Opening Reception	Exhibit Hall
Tue, 8 May 9:00 a.m. - 5:00 p.m. Wed, 9 May, 9:00 a.m. - 12:00 noon	Exhibit	Exhibit Hall
Mon, 7 May 5:00 p.m. - 5:30 p.m.	New Student Orientation	Nicollet C
Mon, 7 May 5:30 p.m. - 7:00 p.m.	Student Meet and Greet	Regency
Tue, 8 May 9:30 a.m. - 11:00 a.m.	Spanish-Speaking Acousticians	St. Croix
Tue, 8 May 1:30 p.m. - 2:30 p.m.	Women in Acoustics Roundtable	Exhibit Hall
Tue, 8 May 2:30 p.m. - 6:00 p.m.	Science Communication Workshop	Mirage
Tue, 8 May 6:00 p.m. - 7:30 p.m.	Social Hour	Exhibit Hall
Wed, 9 May, 11:45 a.m. - 1:45 p.m.	Women in Acoustics Luncheon	Regency
Wed, 9 May, 3:30 p.m.	Annual Membership Meeting	Nicollet B/C
Wed, 9 May, 3:30 p.m. - 5:00 p.m.	Plenary Session/Awards Ceremony	Nicollet B/C
Wed, 9 May, 6:00 p.m. - 8:00 p.m.	Student Reception	Regency
Wed, 9 May, 8:00 p.m. - 12:00 midnight	ASA Jam	Nicollet D
Thu, 10 May, 12:00 noon - 2:00 p.m.	Society Luncheon and Lecture	Regency
Thu, 10 May, 6:00 p.m. - 7:30 p.m.	Social Hour	Exhibit Hall

175th Meeting of the Acoustical Society of America

The 175th meeting of the Acoustical Society of America will be held Monday through Friday, 7–11 May 2018 at the Hyatt Regency Minneapolis Hotel, Minneapolis, Minnesota.

SECTION HEADINGS

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2. TRANSPORTATION AND TRAVEL
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7. TECHNICAL SESSION DESIGNATIONS
8. HOT TOPICS SESSION
9. MEDWIN PRIZE IN ACOUSTICAL OCEANOGRAPHY AND ACOUSTICAL OCEANOGRAPHY PRIZE LECTURE
10. WILILAM AND CHRISTINE PRIZE IN AUDITORY NEUROSCIENCE AND THE AUDITORY NEUROSCIENCE PRIZE LECTURE
11. TUTORIAL LECTURE
12. SHORT COURSE
13. EXHIBIT AND EXHIBIT OPENING RECEPTION
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16. WOMEN IN ACOUSTICS ROUND-TABLE DISCUSSION SESSION
17. ANNUAL MEMBERSHIP MEETING
18. PLENARY SESSION AND AWARDS CEREMONY
19. ANSI STANDARDS COMMITTEES
20. COFFEE BREAKS
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22. PROCEEDINGS OF MEETINGS ON ACOUSTICS
23. E-MAIL AND INTERNET ZONE
24. SOCIALS
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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 Hyatt Regency Minneapolis Hotel is the headquarters hotel where all meeting events will be held.

The cut-off date for reserving rooms at special rates has passed.

Please contact Hyatt Regency Minneapolis, (888-421-1442), 1300 Nicollet Mall, Minneapolis, MN 55403 for information about room availability.

2. TRANSPORTATION AND TRAVEL

The Minneapolis-St Paul International Airport (MSP), is the country's 14th busiest travel hub with 34 million passengers passing through each year. Central location offers a speedy trip (15–30 minutes) to the city.

Transportation options between the Hyatt and the airport include:

Light Rail: The Terminal 1 light rail station is located below the Transit Center, between the Blue and Red ramps. From the Tram Level (one level below bag claim), take the tram to the Transit Center. When you exit the tram, follow the signs to the light rail station. The Terminal 2 light rail station is located on the north side of the Orange Ramp. From Level 1 near Ticketing take the elevator or escalator up to the Orange Ramp skyway. Follow the signs to the LRT station. Take the escalators or elevators down one level to the station platform. The light rail stations are fully accessible.

Blue Line to the Nicollet Avenue stop, the hotel is 8 blocks south on Nicollet Avenue. You can use your Light Rail ticket to take a bus on Nicollet Mall to 13th Street. Fares are $2.00 for Off-Peak times and $2.50 from 6-9 am and 3-6:30 pm.

Taxi: Taxi service at Terminal 1 is accessible via the Tram Level (Level T). Signs direct passengers one level up to the taxi starter booth, where airport staff will assist passengers in obtaining a taxi. At Terminal 2, taxi service is available at the Ground Transport Center, located on the ground level of the Purple Ramp directly across from the terminal building. Downtown Minneapolis is approximately 12 miles from the airport, with fares averaging $39-$49.

Airport Shuttle: Shared ride service to and from the airport is available through SuperShuttle. Ticket counters are located in the Terminal 1-Lindbergh Ground Transport Center, accessible via the terminal's Tram Level. Follow the signs to the escalator or elevator up. Advance reservations are highly recommended. Travelers arriving at Terminal 2-Humphrey can call for pick up at the Super Shuttle kiosk in the Ground Transport Center, located on Level 1 of the Purple Ramp across from the terminal building.

Car Rental: On-airport rental car counters at Terminal 1 are located on the second and third levels between the Blue and Red parking ramps. At Terminal 2, on-airport rental car counters are located in the Ground Transport Center on the ground level of the Purple Ramp directly across from the terminal building.

Hotel valet parking is $38/day, self-parking is $20/day.

3. MESSAGES FOR ATTENDEES

A message board will be located in the Nicollet Promenade 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, 7 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, $250 for Emeritus members (Emeritus status pre-approved by ASA), $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 2018 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.

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 101 sessions with 955 abstracts scheduled for presentation during the meeting.

A floor plan of the Hyatt Regency Minneapolis 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, 7 May
2-Tuesday, 8 May
3-Wednesday, 9 May
4-Thursday, 10 May
5-Friday, 11 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 or Administrative Committee that organized the session using the following abbreviations or codes:

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

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

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

8. HOT TOPICS SESSION

The Hot Topics session (3pID) will be held on Wednesday, 9 May, at 2:15 p.m. in Greenway C/D. Papers will be presented on current topics in the fields of Engineering Acoustics, Signal Processing in Acoustics, and Structural Acoustics and Vibration.

9. MEDWIN PRIZE IN ACOUSTICAL OCEANOGRAPHY AND ACOUSTICAL OCEANOGAPHY PRIZE LECTURE

The 2018 Medwin Prize in Acoustical Oceanography will be presented to Ying-Tsong Lin, Woods Hole Oceanographic Institution, at the Plenary Session on Wednesday, 9 May. Dr. Lin will present the Acoustical Oceanogaphy Prize Lecture on Tuesday, 8 May, at 11:00 a.m. in Session 2aAOb in Greenway A.

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

The 2018 William and Christine Prize in Auditory Neuroscience will be presented to Shihab Shamma, University of Maryland College Park, at the Plenary Session on Wednesday, 9 May. Dr. Shamma will present the Auditory Neuroscience Prize Lecture on Wednesday, 9 May, at 11:00 a.m. in Session 3aPPb in Nicollet D2.

11. TUTORIAL LECTURE ON HEARING LOSS AND THE FUTURE OF AUDITORY IMPLANTS

A tutorial presentation titled "Hearing loss and the future of auditory implants" will be given by Andrew J. Oxenham, University of Minnesota, on Monday, 7 May, at 7:00 p.m. in Nicollet D2.

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).

12. SHORT COURSE ON OPEN SCIENCE AND RELATED TOPICS IN HEARING RESEARCH

A short course on Open Science and Related Topics in Hearing Research will be given in two parts: Sunday, 6 May, from 1:00 p.m. to 5:00 p.m. and Monday, 7 May, from 8:30 a.m. to 12:30 p.m. in the Mirage Room.

The objective of this course is to introduce the various parts of Open Science and to provide researchers related to hearing research with hands-on tools and workflow examples for their everyday use.. Onsite registration at the meeting will be on a space-available basis.

13. EXHIBIT AND EXHIBIT OPENING RECEPTION

An instrument and equipment exhibit conveniently located near the registration area and meeting rooms, will be located in the Exhibit Hall adjacent to Nicollet Promenade.

The Exhibit will include computer-based instrumentation, scientific books, sound level meters, sound intensity systems, signal processing systems, devices for noise control and acoustical materials, active noise control systems and other exhibits on acoustics.

The Exhibit will open with an evening reception on Monday with lite snacks and a complimentary drink. Coffee breaks on Tuesday and Wednesday mornings will be held in the exhibit area as well as an afternoon break on Tuesday.

Exhibit hours are Sunday, 7 May, 5:30 p.m. to 7:00 p.m., Tuesday, 8 May, 9:00 a.m. to 5:00 p.m., and Wednesday, 9 May, 9:00 a.m. to 12:00 noon.

14. GALLERY OF ACOUSTICS

The Technical Committee on Signal Processing in Acoustics will sponsor the 17th 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 of Acoustics will be on display in Nicottet Foyer, Monday to Thursday.

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, respectively.

15. TECHNICAL COMMITTEE OPEN MEETINGS

Technical Committees will hold open meetings on Tuesday, Wednesday, and Thursday at the Hyatt Regency Minneapolis. The schedule and rooms for each Committee meeting are given on page A16.

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

16. WOMEN IN ACOUSTICS ROUND-TABLE DISCUSSION SESSION

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, May 8, in the Exhibit Hall. Discussion topics will include navigating careers in academia, government, and industry; mentoring at all levels; work-life balance; and navigating power differentials in your career. Each table will have a topic leader to 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 tables to discuss a new topic at 2:00 p.m. 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.

17. ANNUAL MEMBERSHIP MEETING

The Annual Membership Meeting of the Acoustical Society of America will be held at 3:30 p.m. on Wednesday, 9 May 2017, in Nicollet B/C at the Hyatt Regency Minneapolis Hotel, 1300 Nicollet Mall, Minneapolis, Minnesota 55403.

18. PLENARY SESSION AND AWARDS CEREMONY

A plenary session will be held Wednesday, 9 May, at 3:30 p.m. in Nicollet B/C.

ASA scholarship recipients and the Student Council Mentor Award recipient will be introduced. The Medwin Prize in Acoustical Oceanography will be presented to Ying-Tsong Lin. The William and Christine Hartmann Prize in Auditory Neuroscience will be presented to Shihab Shamma. The von Békésy Medal will be presented to David Kemp. The R. Bruce Lindsay Award will be presented to Yun Jing. The Helmholtz-Rayleigh Interdisciplinary Silver Medal in Physical Acoustics and Biomedical Acoustics will be presented to Kenneth S. Suslick, and the Gold Medal will be presented to William A. Yost.

Certificates will be presented to Fellows elected at the Boston meeting. See page 1875 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 Minneapolis meeting.

Meetings of selected advisory working groups are often held in conjunction with Society meetings and are listed in the Schedule of Committee Meetings and Other Events on page A16 or on the standards bulletin board in the registration area, e.g., S12/WGI8-Room Criteria.

People interested in attending and in becoming involved in working group activities should contact the ASA Standards Manager for further information about these groups, or about the ASA Standards Program in general, at the following address: Neil Stremmel, 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 9:30 a.m. to 11:00 a.m. The breaks on Monday, Thursday, and Friday will be held in Nicollet Foyer. The breaks on Tuesday and Wednesday will be held in the Exhibit Hall. There will also be an afternoon break on Tuesday from 2:30 p.m. to 3:45 p.m. in the Exhibit Hall.

21. A/V PREVIEW ROOM

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

22. PROCEEDINGS OF MEETINGS ON ACOUSTICS (POMA)

The Minneapolis 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 first floor in Nicollet Promenade.

Tables with power cords will be set up in Nicollet Promenade 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 the Exhibit Hall.

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, 10 May, at 12:00 noon in the Regency Room. The program will be presented by members of Cantus, the acclaimed men's vocal ensemble (cantussings.org). Cantus has been lauded as the premier men's vocal ensemble in the United States. They are committed to ensuring the future of ensemble singing by mentoring young singers and educators. Since the very early years, Cantus has been offering master classes and lectures

This luncheon is open to all attendees and their guests Purchase your tickets at the Registration Desk before 10:00 a.m. on Wednesday, 4 December. 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, 7 May, from 5:00 p.m. to 5:30 p.m. in Nicollet C. This will be followed by the Student Meet and Greet from 5:30 p.m. to 6:45 p.m. in Greenway F/G where refreshments and a cash bar will be available.

The Students' Reception will be held on Wednesday, 9 May, from 6:00 p.m. to 8:00 p.m. in the Regency 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 11:30 a.m. on Wednesday, 9 May, in the Regency Room. Those who wish to attend must purchase their tickets in advance by 10:00 a.m. on Tuesday, 8 May. The fee is USD $30 for non-students and USD$15 for students.

28. JAM SESSION

You are invited on Wednesday night, 5 December, from 8:00 p.m. to midnight in Nicollet D for the ASA Jam. 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 Minneapolis meeting. The on-site registration fee for accompanying persons is USD $200. A hospitality room for accompanying persons will be open in the Regency Room from 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 the meeting website for updated information.

Minneapolis is a city rich in music, architecture and cultural institutions, as well as a hub for sports and shopping. Within a short walk of the hotel are dozens of museums and landmarks as well as a broad range of culinary experiences. There are also a variety of excellent tours to choose from.

30. WEATHER

In Minneapolis, the month of May is characterized by rapidly rising daily high temperatures, with daily highs increasing by 10°F, from 64°F to 74°F over the course of the month, and rarely exceeding 85°F or dropping below 51°F. The average high and low temperatures are 66°F and 46°F, respectively. The probability of rainfall is 32% during the beginning of May.

31. TECHNICAL PROGRAM ORGANIZING COMMITTEE

Andrew J. Oxenham, Chair; David P. Knobles, Acoustical Oceanography; David Mann, Animal Bioacoustics; Ian Hoffman, Shane Kanter, Architectural Acoustics; Siddhartha Sikdar, Kang Kim, Subha Maruvada, Biomedical Acoustics; Daniel Russell, Education in Acoustics; Kenneth M. Walsh, Engineering Acoustics; Whitney Coyle and Peter Rucz, Musical Acoustics; William J. Murphy, Noise, Michael Haberman and Kevin Lee, Physical Acoustics; Anna Diedesch, Psychological and Physiological Acoustics; Buye Xu, Ballard Blair, Signal Processing in Acoustics; Melissa Baese-Berk, Kristin Van Engen, Speech Communication; Benjamin Shafer, Robert A. Koch, Structural Acoustics and Vibration; Julien Bonnel, Underwater Acoustics; Kelly Whiteford, Vahid Naderyan, Jonathan Weber, Student Council.

32. MEETING ORGANIZING COMMITTEE

Peggy B. Nelson and Bruce C. Olson, Cochairs; Andrew J. Oxenham, Technical Program Chair; Benjamin Munson and Adam Svec, Student Coordinators; Matt Hildebrand, Signs; Alan Nelson and Milton Salcedo, Accompanying Persons.

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 2018 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 CD or 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 1024x768 pixels or to the resolution indicated by the AV technical support. If it looks OK, it will probably look OK to your audience during your presentation.
- Remember that graphics can be animated or quickly toggled among several options: Comparisons between figures may be made temporally rather than spatially.
- Animations often run more slowly on laptops connected to computer video projectors than when not so connected. Test the effectiveness of your animations before your assigned presentation time on a similar projection system (e.g., in the A/V preview room). Avoid real-time calculations in favor of pre-calculation and saving of images.
- If you will use your own laptop instead of the computer provided, connect your laptop to the projector during the question/answer period of the previous speaker. It is good protocol to initiate your slide show (e.g., run PowerPoint) immediately once connected, so the audience doesn't have to wait. If there are any problems, the session chair will endeavor to assist you, but it is your responsibility to ensure that the technical details have been worked out ahead of time.
- During the presentation have your laptop running with main power instead of using battery power to insure that the laptop is running at full CPU speed. This will also guarantee that your laptop does not run out of power during your presentation.

Specific Hardware Configurations

Macintosh

Older Macs require a special adapter to connect the video output port to the standard 15-pin male DIN connector. Make sure you have one with you.

- Hook everything up before powering anything on. (Connect the computer to the RGB input on the projector).
- Turn the projector on and boot up the Macintosh. If this doesn't work immediately, you should make sure that your monitor resolution is set to 1024x768 for an XGA projector or at least 640x480 for an older VGA projector. (1024x768 will most always work.). You should also make sure that your monitor controls are set to mirroring. If it's an older PowerBook, it may not have video mirroring, but something called simulscan, which is essentially the same.
- Depending upon the vintage of your Mac, you may have to reboot once it is connected to the computer projector or switcher. Hint: you can reboot while connected to the computer projector in the A/V preview room in advance of your presentation, then put your computer to sleep. Macs thus booted will retain the memory of this connection when awakened from sleep.
- Depending upon the vintage of your system software, you may find that the default video mode is a side-by-side configuration of monitor windows (the test for this will be that you see no menus or cursor on your desktop; the cursor will slide from the projected image onto your laptop's screen as it is moved). Go to Control Panels, Monitors, configuration, and drag the larger window onto the smaller one. This produces a mirror-image of the projected image on your laptop's screen.
- Also depending upon your system software, either the Control Panels will automatically detect the video projector's resolution and frame rate, or you will have to set it manually. If it is not set at a commensurable resolution, the projector may not show an image. Experiment ahead of time with resolution and color depth settings in the A/V preview room (please don't waste valuable time adjusting the Control Panel settings during your allotted session time).

PC

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

Linux

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

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

176th Meeting, Victoria, Canada, 6–9 November 2018

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

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

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

180th Meeting, Cancun, Mexico, fall 2020

FIFTY-YEAR AWARDS

The following individuals have been members of the Society for fifty years. They will receive "Gold" certificates in recognition of their continuing interest in the Society for half a century.

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The following individuals have been members of the Society for twenty-five years. They will be sent "Silver" certificates in recognition of the mutual advantages derived from their long-time association with the Society.

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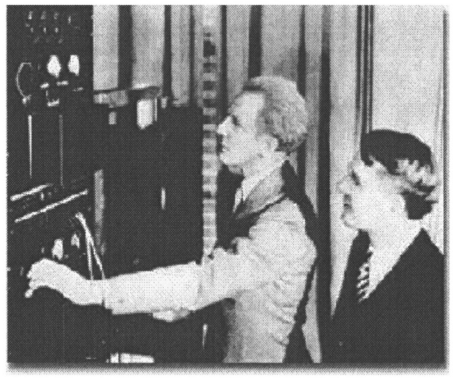
Study of Speech and Hearing at Bell Telephone Laboratories: Correspondence Files (1917-1933) and Other Internal Reports

During the first half of the 20th century. Bell Telephone Laboratories (BTL) carried out an extensive research program aimed at improving the quality of telephone-transmitted speech. This effort was led by Harvey Fletcher who also served as the first president of the Acoustical Society of America. The contributions of Fletcher and his colleagues helped produce not only the world's leading telephone system but also built the foundation of the field of speech and hearing sciences.

This compact disc (CD) contains nearly 10,000 pages of internal documents generated at BTL during that era. The majority are scientific reports showing the extraordinary thinking and foresight of these world-class reseachers. In addition to their historical importance, the documents also provide insight and clarity on topics of current interest to speech and hearing scientists.



Three categories of documents are included on the CD. **Correspondence Files** include notes and letters sent or received by administrators and researchers ranging on topics from equipment requisitions to discussions of project plans and experimental results. **Historical Documents** include technical reports that were filed separately from the speech and hearing correspondence files. **Laboratory Notebooks** of R. Galt are included because of their relevance comprehensibility.
The CD was compiled by Christine M. Rankovic and Jont B. Allen of AT&T Bell Laboratories in Murray Hill, New Jersey.

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Session 1aAO

Acoustical Oceanography and Underwater Acoustics: Acoustics in Naturally Constrained Environments: Estuaries, Bays, Inlets, Fjords, and Rivers I

Andone C. Lavery, Cochair
Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, 98 Water Street, MS 11, Bigelow 211, Woods Hole, MA 02536

D. Benjamin Reeder, Cochair
Oceanography, Naval Postgraduate School, 833 Dyer Rd., Bldg. 232, SP-311B, Monterey, CA 93943

Chair's Introduction—9:30

Invited Papers

9:35

1aAO1. Broadband acoustic wave propagation and scattering in Delaware Bay Estuary. Mohsen Badiey (Univ. of Delaware, Newark, DE 19716, badiey@udel.edu)

Physical oceanography of estuaries is complex due to many factors including fresh water river discharge, oceanic salt water intrusion, variable ebb and flood tides, and surface winds. In these confined environments, high frequency (1–25 kHz) broadband acoustic waves are shown to through large intensity fluctuations with substantial energy scattering due to their interaction with the sea surface, sea bottom, and refraction due to the stratified water column. In this paper, we present results from a series of experiments that were conducted in the Delaware Bay estuary where fixed source-receiver configurations allowed calibrated long-term acoustic and oceanographic measurements to be conducted while atmospheric conditions such as wind speed and direction were carefully measured concurrently. It is shown that intensity fading due to oceanographic features and tide induced sound speed profile refraction in stratified water column occurs regularly and periodically. Wind generated surface gravity water waves in a fetch limited environment also play an important role in modulating and fading surface bounced acoustic energy. Two dimensional Parabolic Equation (PE) model with rough surface is used to simulate the measured broadband signals. Excellent data-model comparison is shown. [Work supported by ONR321OA.]

9:55

1aAO2. Three-dimensional modeling in the Weymouth Fore River. Michael B. Porter (HLS Res., 12625 High Bluff Dr., Ste 211, San Diego, CA 92130, mikeporter@hlsresearch.com), Timothy F. Duda, Peter Traykovski, Arthur Newhall (Woods Hole Oceanographic Inst., Woods Hole, MA), and Laurel Henderson (HLS Res., La Jolla, CA)

The Weymouth Fore River in Eastern Massachusetts is an estuary that presents an interesting example of a constrained environment. The river bank, of course, channels the energy. Man-made features such as the Fore River Bridge, several piers, and the USS Salem (a heavy cruiser) also present interesting features in terms of the sound propagation. We are using this site as the first of several to validate acoustic modeling tools in such constrained spaces. A detailed bathymetric survey was done providing centimeter level resolution using the Woods Hole Jetyaks. The environmental information is the input to a three-dimensional beam tracing model. This talk will present preliminary comparisons of the measured and modeled acoustic data.

10:15

1aAO3. A propagation experiment in an Arctic glacial fjord. Grant B. Deane (Marine Physical Lab., Univ. of California, San Diego, 13003 Slack St., La Jolla, CA 92093-0238, gdeane@ucsd.edu) and Oskar Glowacki (Polar Sci., Inst. of Geophys., Warsaw, Poland)

A propagation experiment was undertaken in the meltwater-modified surface layer in the bay of a marine-terminating Arctic glacier. Broad-band, m-sequence transmissions were made with an International Transducer Corporation 1007 source deployed from a drifting boat tracked using gps, approximately 500 m in front of Hansbreen Glacier in Hornsund Fjord, Southwestern Svalbard in the summer of 2017. Signals were received with a short array of Hitech HTI-96 hydrophones tethered to an autonomous, anchored boat. The distance between the hydrophone array and source was measured every 2 minutes using a laser rangefinder. Propagation effects in this noisy and dynamic environment include waveguide propagation through the meltwater-modified surface layer and occlusion of transmissions by drifting icebergs. Details of the experiment and propagation effects noted during the transmission period will be presented and discussed. [Work funded by ONR, Grant No. N00014-17-1-2633, and by the Polish National Science Centre, Grant No. 2013/11/N/ST10/01729.]

10:35–10:50 Break

10:50

1aAO4. Spatio-temporal patterns in underwater sound within an urban estuarine river. Sarah A. Marley (Ctr. for Marine Sci. and Technol. (CMST), Curtin Univ., GPO Box U1987, Perth, WA 6845, Australia, sarah.marley86@gmail.com), Christine Erbe (Ctr. for Marine Sci. and Technol. (CMST), Curtin Univ., Bentley, WA, Australia), Chandra P. Salgado Kent, Miles Parsons, and Iain M. Parnum (Ctr. for Marine Sci. and Technol. (CMST), Curtin Univ., Perth, WA, Australia)

Underwater noise environments are increasingly being considered in marine spatial planning and habitat quality assessments, particularly with regard to acoustically specialised fauna. The Swan River in Western Australia flows through the state capital of Perth and consequently experiences a range of anthropogenic activities. However, the river is also extensively used by a resident community of bottlenose dolphins (*Tursiops aduncus*). This study aimed to describe underwater sound sources within the Swan River, examine spatial and temporal soundscape variability, and determine dolphin responses to noisy environments. Acoustic datasets collected from 2005 to 2015 indicated that the Swan River was comprised of multiple acoustic habitats, each with its own characteristic soundscape and temporal patterns in underwater noise. The anthropogenically "noisiest" site was the Fremantle Inner Harbour (mean broadband noise level: 106 dB re 1 μPa rms [10 Hz–11 kHz]); yet dolphins remained present in this area even at high vessel densities. However, fine-scale analyses indicated significant alterations to dolphin behavior at high vessel densities and to dolphin whistle characteristics in high broadband noise conditions. These results highlight the need to consider spatial and temporal patterns when assessing the composition of underwater soundscapes, and identify potential responses of coastal dolphins to busy, noisy environments.

Contributed Papers

11:10

1aAO5. Underwater sound pressure and particle motion (acceleration, velocity, and displacement) from recreational swimmers, divers, surfers, and kayakers. Christine Erbe (Ctr. for Marine Sci. & Technol., Curtin Univ., Kent St., Bentley, WA 6102, Australia, c.erbe@curtin.edu.au), Miles Parsons, Alec J. Duncan, Klaus Lucke, Alexander Gavrilov, and Kim Allen (Ctr. for Marine Sci. & Technol., Curtin Univ., Perth, WA, Australia)

Naturally constrained environments like bays and lakes are frequently used for water sports activities. While the sound of motorized vessels is rather well understood, non-motorized activities have received less investigation. Ten water sports activities (swimming backstroke, breaststroke, butterfly, and freestyle; snorkelling with fins; kicking a boogie board with fins; paddling with alternating or simultaneous arms while lying on a surfboard; scuba-diving; kayaking and jumping into the water) were recorded in a controlled yet even more constrained environment: an Olympic-sized pool. Activities that occurred at the surface involved repeatedly piercing the surface and hence creating the bubble clouds which were the strongest sound generators. Received levels were 110–131 dB re 1μPa (10–16,000 Hz) for all of the activities at the closest point of approach (1 m). All activities produced a characteristic spectro-temporal pattern in all the quantities measured (sound pressure, particle displacement, velocity, and acceleration) by which they could be identified. Applicability of the results to security monitoring of pools, performance assessments of professional or competitive swimmers, and studies of the distances at which humans may be detectable by marine animals in the sea will be discussed.

11:25

1aAO6. Compressional and shear *in situ* measurements in the Lower Laguna Madre. Justin T. Dubin, Megan S. Ballard, Kevin M. Lee, Andrew R. McNeese, Jason D. Sagers, Gabriel R. Venegas (Dept. of Mech. Eng. and Appl. Res. Labs, The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, justin.dubin@utexas.edu), Abdullah F. Rahman (The University of Texas Rio Grande Valley, Sch. of Earth, Env., and Marine Sci., Brownsville, TX), and Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs, The Univ. of Texas at Austin, Austin, TX)

This presentation describes the apparatus, procedure, and results from a field experiment conducted in July 2017 in the seagrass beds of the Lower Laguna Madre. The Laguna Madre is a shallow hypersaline lagoon on the western coast of the Gulf of Mexico. The relatively flat bathymetry is covered by a thick meadow of *Thalassia testudinum*. Acoustic piezoelectric transducer probes measured compressional wave speed and attenuation in the seagrass canopy as well as geoacoustic properties (compressional and shear wave speed and attenuation) of the underlying sediment. Geoacoustic parameters were measured at depths between 5 cm and 20 cm in 5 cm increments. Compressional wave measurements were made over a frequency range of 20 kHz to 100 kHz, and shear wave measurements were made at 500 Hz and 1 kHz. The compressional wave data from this experiment indicates dispersion within the seagrass canopy and sediment. This dispersion is likely due to the high presence of gas bubbles in both the seagrass and supporting sediment. Shear wave velocity monotonically increases with depth at both frequencies. The measured wave speeds and attenuations are compared to results from sediment-acoustic models. Challenges in data collection and processing are explored. [Work sponsored by ARL:UT IR&D and ONR.]

11:40

1aAO7. Measurements and modeling of acoustic propagation in a seagrass meadow. Megan S. Ballard, Jason D. Sagers, Kevin M. Lee, Andrew R. McNeese (Appl. Res. Labs. at the Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu), Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Abdullah F. Rahman (Sch. of Earth, Env., and Marine Sci., The Univ. of Texas Rio Grande Valley, Brownsville, TX), Justin T. Dubin, and Gabriel R. Venegas (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

This talk presents results from an acoustic propagation experiment conducted in the Lower Laguna Madre to characterize the acoustical properties of a seagrass meadow. At the location of the experiment, the water was one meter deep, and the seabed was covered by a dense growth of *Thalassia testudinum*, a type of seagrass that grows from a long, jointed rhizome buried 5 cm to 10 cm below the seafloor. The biological processes and physical characteristics associated with seagrass are known to affect acoustic propagation due to bubble production, which results in dispersion, absorption, and scattering of sound. During the experiment, a Combustive Sound Source was used to produce broadband signals at ranges of 20 m to 1000 m from the receiver location. Three sensors were positioned at the receiver location: two hydrophones located within and above the seagrass canopy, and a single-axis geophone. The data were analyzed for the purposes of predicting acoustic propagation in seagrass meadows and for estimating environmental parameters in very shallow, biologically active environments. Additionally, seagrasses are a vital part of the global carbon cycle and this work explores the feasibility of using of acoustic signals to estimate carbon stores. [Work sponsored by ARL:UT IR&D and ONR.]

Session 1aNS

Noise, Architectural Acoustics, Physical Acoustics, and Structural Acoustics and Vibration: Novel Materials for Sound Absorption, Insulation, and Vibration Control

Yun Jing, Cochair
North Carolina State University, 911 Oval Dr., EB III, Campus Box 7910, Raleigh, NC 27695

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

Chair's Introduction—8:30

Invited Papers

8:35

1aNS1. Exploration of acoustical and dynamical properties of organic nanoparticle assemblies for increased sound insulation. Mathew A. Whitney, Ning Xiang (Graduate Program in Architectural Acoust., School of Architecture, Rensselaer Polytechnic Inst., Troy, NY 12180, mathew.whitney@gmail.com), Sadeq Malakooti (Mech. Eng., the Univ. of Texas in Dallas, Richardson, TX), Habel G. Churu (Mech. Eng., LeTourneau Univ., Longview, TX), Nicholas Leventis (Chemistry Dept., Missouri Univ. of Sci. and Technol., Rolla, MO), and Hongbin Lu (Mech. Eng., the Univ. of Texas in Dallas, Richardson, TX)

Low-density highly porous solid monolithic macroscopic objects, consisting of hierarchical mesoporous three-dimensional assemblies of nanoparticles (aerogels), have been previously pursued mainly for their low thermal conductivity. Unlike classical aerogels based on silica being fragile materials, structural fragility issue has been addressed with materials referred to as polymer-crosslinked (or X-) aerogels. X-aerogels are low-density materials, yet their mechanical strengths have been significantly increased. This work explores ductile aerogels in potential uses as constrained damping layers integrated into classical wallboards for drastically increased sound transmission losses without significantly increasing the board thickness and weight. Their ductility and mechanical strength enables light-weight X-aerogel panels of less than 1 cm in thickness to be integrated into gypsum wallboards. This work has experimentally investigated the integrated wallboards to achieve significantly increased sound transmission loss without significantly increasing thickness and weight of integrated wallboard system. This paper discusses preliminary investigations on experimental methods to characterize broad-band dynamic properties of X-aerogels for better understanding of its excellent effect in drastically increased sound transmission loss. Some preliminary test results carried out in chamber-based random-incident measurements demonstrate high potentials in building acoustics applications and beyond.

8:55

1aNS2. Large scale metasurfaces for seismic waves control. Antonio Palermo (DICAM, Univ. of Bologna, Viale Del Risorgimento 2, Bologna 40132, Italy, antonio.palermo6@unibo.it), Sebastian Krödel (Dept. of Mech. and Process Eng., ETH, Zurich, Switzerland), Kathryn Matlack (Dept. of Mech. Sci. and Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), Rachele Zaccherini, Vasilis Dertimanis, Eleni Chatzi (Dept. of Civil, Environ. and Geomatic Eng., ETH, Zurich, Switzerland), Alessandro Marzani (DICAM, Univ. of Bologna, Bologna, Europe, Italy), and Chiara Daraio (Div. of Eng. and Appl. Sci., California Inst. of Technol., Pasadena, CA)

Elastic metamaterials are artificial composites with subwavelength resonant particles hosted in a medium able to manipulate the propagation of elastic waves. When the resonant particles are placed at the free surface of the medium to form a resonant "metasurface," the localization mechanism and the direction of surface waves can be fully controlled. In this talk, we discuss the use of resonant metasurfaces to control the propagation of vertically and horizontally polarized surface waves and their possible application for seismic waves mitigation. By combining analytical, numerical, and experimental studies, we describe the interaction of Rayleigh waves with a metasurface of vertical resonators and design large-scale resonant barriers to deviate damaging seismic Rayleigh waves into the medium bulk. Additionally, we investigate the effect of material stratification on the metasurface dynamics by analyzing the propagation of surface waves in unconsolidated granular media with depth-dependent stiffness profile. Finally, we describe the interaction of Love waves guided by a stratified medium with a metasurface of horizontal resonators and design large-scale resonant metalenses to redirect their propagation.

9:15

1aNS3. Fractal and bio-inspired labyrinthine acoustic metamaterials. Anastasiia Krushynska, Federico Bosia (Dept. of Phys., Univ. of Turin, Via P. Guiria 1, Turin 10125, Italy, akrushynska@gmail.com), and Nicola M. Pugno (Dept. of Civil, Environ. and Mech. Eng., Univ. of Trento, Trento, Trentino, Italy)

Control of low-frequency sound is a challenge, despite numerous advances in the field. Recently emerged acoustic metamaterials have already proven their efficiency for the development of innovative systems for sound control and demonstration of such fascinating phenomena as super-resolution imaging, transformation acoustics, acoustic cloaking, etc. For sound attenuation, a common attenuation mechanism relies on dispersion induced by local resonances that results in the generation of slow sound inside a metastructure at resonant frequencies. However, the selective bandwidth of metamaterials with identical resonators restricts their practical applications. To overcome this limitation, one can introduce rainbow-type resonators that leads to the increase of structural sizes and raises manufacturing costs. We present fractal and bio-inspired labyrinthine acoustic metamaterials with non-resonant elements for low-frequency sound control. Designed configurations contain sets of narrow channels extending wave propagation paths that leads to reduction of effective sound speed within a structure. We show that by folding these channels into fractal space-filling curves or bio-inspired geometries, one can induce strong artificial Mie-type resonances that allow the achievement of total broadband sound reflection at deep subwavelength scales. This unique phenomenon can be used to develop small-size metastructures with total reflectance of low-frequency sound for versatile applications.

Contributed Papers

9:35

1aNS4. Honeycomb metasurface panel for deep-subwavelength broadband sound absorption. Xiuyuan Peng (Nanjing Univ., Nanjing, China), Yuanchen Deng, and Yun Jing (North Carolina State Univ., 911 Oval Dr., EB III, Campus Box 7910, Raleigh, NC 27606, ydeng5@ncsu.edu)

Honeycomb sandwich panels have long been preferred for their good mechanical properties. Here, we demonstrate a design of honeycomb metasurface panel that can achieve 90% sound absorption from 600 Hz to 1000 Hz with a thickness of less than 30 mm (approximately $\lambda/20$) by applying minor modifications to commercialized honeycomb sandwich panels. The panel was made up of periodically and horizontally arranged honeycomb "supercells" which consist of different types of honeycomb unit cells. Each unit cell could be considered a Helmholtz resonator (HR) with hexagonal prism-shaped cavity and cylindrical neck. The absorption performance of designed honeycomb metasurface panel was studied both with surface impedance theory and the complex frequency plane theory, and was then validated by numerical simulations.

9:50

1aNS5. Ultrathin Ashoka Chakra like acoustic metastructure as a sound absorber. Sanjay Kumar (Dept. of Mech. Eng., Indian Inst. of Technol. Kanpur, Microsystems Fabrication Lab., IIT Kanpur, Kanpur, Uttar Pradesh 208016, India, sanjay21505@gmail.com), Pulak Bhushan (Dept. of Elec. and Comput. Eng., Florida Int. Univ., Miami, FL), and Shantanu Bhattacharya (Mech. Eng., Indian Inst. of Technol. Kanpur, KANPUR, India)

Absorption of undesirable sound is a fundamental issue with broad applications in a variety of situations, such as in a conference room, cinema hall, aircraft cabin noise, etc. A variety of sound absorbers, e.g., micro-slit, concrete walls, porous media, etc., are used for noise control but are required to have a thickness comparable to the working wavelength. Here, we report a novel ultrathin "Ashoka Chakra" like acoustic metastructure that can be utilized as a sound absorber. The developed acoustic metastructure demonstrates broad bandwidth and high absorption characteristics. Each unit cell contains 24 hollow spokes through which acoustic wavefront can move freely. Despite its simplicity, this approach provides tunability of the metastructure functionality such as operating frequency range by altering the structure dimensions. Finite element based simulations were carried out in order to optimize the metastructure's dimensions to achieve maximum absorption. The efficacy of the metastructure is validated through excellent agreement between simulation and experimental measurements. Realization of the proposed acoustic metastructure can show promising applications in the field of sound absorption.

10:05–10:20 Break

10:20

1aNS6. 3D printed sound absorbers using functionally-graded sonic crystals. Matthew D. Guild (Acoust. Div., Naval Res. Lab., Code 7165, 4555 Overlook Ave., SW, Washington, DC 20375, matthew.guild@nrl.navy.mil), Michael Rothko (Dept. of Mech. Eng., Boston Univ., Boston, MA), Caleb F. Sieck (NRC Postdoctoral Res. Program, Naval Res. Lab, Austin, Texas), Charles Rohde, and Gregory Orris (Acoust. Div., Naval Res. Lab., Washington, DC)

In this work, a thin functionally-graded sound absorber that achieves an absorption coefficient near unity is demonstrated. The sound absorber consists of a multilayer arrangement of an interwoven sonic crystal lattice with varying filling fractions, backed by a thin elastic coating that acts as a flexural acoustic element. The overall thickness of the sound absorber is about one tenth of the wavelength in air, and it was 3D printed with a thermoplastic polyurethane. Samples were fabricated and acoustically tested in an air-filled acoustic impedance tube, from which absorption and effective acoustic properties were obtained. A theoretical formulation for the effective acoustic properties of the sonic crystal lattice was used to guide the design process, and excellent agreement was found between measured and theoretically predicted results. A range of sonic crystal filling fractions and thicknesses were tested to verify the fabrication process and robustness of the theoretical formulation, and both were found to be in excellent agreement. Based on this testing and analysis, an optimal arrangement was found that achieved simultaneous near zero reflectance and transmittance over a given frequency band, thereby resulting in an absorption coefficient near unity. [Work supported by the Office of Naval Research.]

10:35

1aNS7. Modeling acoustic resonators: An application to resonator-enclosure coupling. Caleb B. Goates, Scott D. Sommerfeldt (Brigham Young Univ., N283 ESC, Provo, UT 84602, calebgoates@gmail.com), and David C. Copley (Caterpillar, Inc., Peoria, IL)

Acoustic resonators, such as the Helmholtz resonator, provide a stable, cost effective passive noise control solution, and have been widely used to attenuate unwanted sound in enclosures and ducts. Classical formulations predicting the input impedance of such resonators often have significant error, creating a need for repeated prototyping or tuning during fabrication to achieve the desired response. Previous work found that higher-order calculations, including impedance translation and equivalent circuit modelling, produce much more accurate predictions [Calton and Sommerfeldt, J. Acoust. Soc. Am. **139**, 2205 (2016)], allowing prototyping to be done quickly on a computer before fabrication of the resonators. This talk will continue the discussion in the work referenced above. In addition to resonator impedance predictions, resonator-enclosure coupling predictions will be discussed. It will be shown through comparison of predicted and experimental results that the impedance and coupling predictions can remove the need for repeated prototyping and tuning. We will also show the incorporation of

these predictions into a user interface allowing engineers without acoustics background the ability to design resonators for passive noise control.

10:50

1aNS8. Fibrous material microstructure design for optimal structural damping. Yutong Xue and J. S. Bolton (Ray W. Herrick Labs., School of Mech. Eng., Purdue, 177 S. Russell St., West Lafayette, IN 47906, xue46@purdue.edu)

It is shown here that properly designed fibrous media (e.g., glass or polymeric fibers) can be used to damp structural vibration as well as to absorb sound. The materials can then be multi-functional, reducing the number of elements required to achieve a specified level of noise and vibration performance. Past work demonstrated that layers of fibrous media placed on panels can damp the panel motion by removing energy from the nearfield acoustical motion generated by the panel vibration. The current study focused on designing the fibrous medium to ensure optimal vibration damping in a particular application. First, a method of calculating the response of a panel with an attached fibrous layer is recalled and updated to allow layers of limp or elastic porous media to be modeled. Example results will be presented, and then it will be shown that an optimal flow resistivity exists for a given frequency range and configuration of interest. Finally, based on a recently developed theory for the flow resistivity of fibrous media, the optimal flow resistivity identified in that way can be translated into a particular fiber size, given properties such as density of the fiber material and the desired superficial density.

11:05

1aNS9. Groundborne vibration attenuation by particulate dampers to reduce building vibration at sonic frequencies. Hasson M. Tavossi (Dept. of Eng. Technol., Savannah State Univ., Box 20089 3219 College St., Savannah, GA 31404, tavossih@savannahstate.edu)

Particulate medium can act as efficient damper for building vibrations. Mechanical properties of particulate material and their size distributions determine their vibration absorption characteristics. Unconsolidated particulate damper of uniform size is shown to have bandpass filter behavior for mechanical vibrations that are transmitted through them. The dampers absorption spectrum also shows bandgap structure where mechanical vibration in each frequency range is strongly absorbed. We use samples of uniform size spherical particulates that are subjected to mechanical vibrations in the audible frequency range to determine their absorption characteristics as a function of particle size, frequency, intensity, and direction of propagation. The effects of particle sizes and layer thickness on vibration absorption are determined. The goal is to determine the required absorption characteristics as a function of particle size distribution, that is required to mitigate building vibrations.

11:20

1aNS10. Vibration isolation of sensory deprivation tanks. Erik Miller-Klein (A3 Acoust., 241 South Lander St, Ste. 200, Seattle, WA 98134, erik@a3acoustics.com) and Matthew V. Golden (Pliteq, Washington, DC)

Sensory deprivation tanks or float tanks are advertised to be a powerful stress-relief and wellness tool. In these tanks, a person floats in a salt water bath and is enclosed to cut off all visual and auditory sensory inputs. Vibration can have a significant effect on the experience of using a one of these tanks by disrupting the sensory deprivation experience. A recent project will be discussed where vibration was causing such a problem. A solution was developed using recycled rebounded rubber crumb. The subject and object performance data will be shared.

11:35

1aNS11. Improving traffic noise transmission loss of plenum windows by using sound scatterer arrays. Shiu-Keung Tang (Dept. of Bldg. Services Eng., The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong, shiu-keung.tang@polyu.edu.hk)

Plenum windows have been used recently in residential buildings erected next to busy trunk roads. These windows are able to provide an additional 8 dB traffic noise reduction than the conventional casement windows, but at the same time they can allow for natural ventilation of the residential units they protect acoustically. The window design consists of two parallel glass panes—one at the building façade (outer) and the other in the indoor side (inner). The outer and inner window openings are located on opposite window of the window span so that they, together with the cavity between the two window panes, form a passage for air movement, while the window's structure acts as an acoustic filter to attenuate noise. To improve the acoustical performance of this window design, it is proposed in this study to install sound scatterer arrays into the window cavity. It is found from finite-element simulation that such proposal, under appropriate settings, can result in about 4–5 dB improvement of traffic noise transmission loss across a plenum window without significant reduction of air movement across the window. The acoustical performance will further be improved without the increase in air flow resistance when the scatterer rows are staggered.

Session 1aPA

Physical Acoustics, Structural Acoustics and Vibration, Architectural Acoustics, and Underwater Acoustics: Novel Methods in Computational Acoustics I

D. Keith Wilson, Cochair
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Amanda Hanford, Cochair
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Chair's Introduction—8:40

Invited Papers

8:45

1aPA1. Deep reinforcement learning for cognitive sonar. Jason E. Summers (ARiA, 1222 4th St. SW, Washington, DC 20024-2302, jason.e.summers@ariacoustics.com), Jason M. Trader (ARiA, Culpeper, VA), Charles F. Gaumond (ARiA, Washington, DC), and Johnny L. Chen (ARiA, Culpeper, VA)

Current developments in cognitive sonar have leveraged explicit semantic representations such as ontologies to develop cognitive agents that enable adaptation of sonar settings to optimize behaviors such as waveform selection during active tracking. However, such cognitive systems based on explicit knowledge representations can be time-and-labor intensive to update in operation and limited in ultimate performance. In other applications, such as computer Go, breakthrough performance was achieved by going beyond learning from all prior games and actions of experts to allowing the algorithm to compete with itself to develop and learn from the outcome of new approaches. This hybrid approach of learning from experts and then learning via self-competition is vital for sonar applications such as active antisubmarine warfare (ASW) because ground-truthed performance data are sparse and may not display optimal solutions. Here, we discuss our application of reinforcement learning to active ASW in a simulated environment based on computational-acoustics models, including an assessment of the goal states and performance metrics we have used. The fidelity of the simulated environment is discussed in terms of the interaction with reinforcement learning and the impact on generalization from learning in simulated environments to application in real environments. [Work supported by ARiA IR&D.]

9:05

1aPA2. Machine learning for spatial interpolation of noise monitor levels. Edward T. Nykaza and Matthew G. Blevins (ERDC-CERL, 2902 Newmark Dr., Champaign, IL 61822, edward.t.nykaza@usace.army.mil)

Continuously recording noise monitoring stations provide feedback of the noise environment at monitor locations. While this feedback is useful, it only provides information at a few point locations and, in many cases, it is of interest to know the noise level(s) at locations between and beyond noise monitoring locations. In this study, we test several machine learning models (e.g., random forests, support vector machines, and neural networks) for their ability to predict noise levels using a small number of noise monitors. Unlike the hybrid geostatistical-acoustical method proposed by Nykaza [Nykaza, Ph.D. thesis, 2013] which requires knowledge of the source location, the methods proposed in this study can be trained to estimate noise levels without knowledge of the source location. The accuracy of each model is assessed with the root-mean-square-error, and we discuss the practical implications of implementing such approaches in real-time systems.

9:25

1aPA3. Coupled FE/BE for periodic acoustic systems. Andrew S. Wixom, Amanda Hanford, Jonathan S. Pitt, and Douglas E. Wolfe (Appl. Res. Lab., Penn State Univ., P.O Box 30, M.S. 3220B, State College, PA 16801, axw274@psu.edu)

While coupled finite element and boundary element (FE/BE) codes are used regularly in acoustics—particularly structural acoustics—and periodic boundary conditions are common, the combination of the three is rare. However, such calculations have been performed in the electricity and magnetism community for quite some time. This talk presents a fully coupled FE/BE framework for acoustics with doubly-periodic boundary conditions, accomplished by way of the Ewald transformation of the appropriate periodic Green's function. This method is likely of primary interest to the acoustic metamaterials community and it is used to analyze an example problem drawn from this field. Both the internal acoustic field within the FE mesh as well as the system's radiation characteristics are examined.

9:45

1aPA4. Verification and validation of a coupled elasto-acoustic-damage system. Jonathan S. Pitt (The Penn State Univ., PO Box 30, State College, PA 16804-0030, jsp203@psu.edu)

A novel time-domain method for simulating dynamic damage evolution in a coupled structural-acoustic system is presented. The system is derived via the theory of continuum damage mechanics, and incorporates standard damage evolution models, but is readily extendible to more complex formulations. 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 are presented, and present a novel first principles study of changes in the structural acoustic response to dynamically evolving damage in the structure. Special attention is given to brittle fracture. Examples of downstream usage of the evolving structural response are discussed in the conclusion.

10:05

1aPA5. Numerical investigation of coupling schemes for structural acoustics. Scott T. Miller, Gregory Bunting (Computational Solid Mech. & Structural Dynam., Sandia National Labs., P.O. Box 5800, MS 0845, Albuquerque, NM 87185, stmille@sandia.gov), and Nicholas A. Reynolds (Code 6605, Naval Surface Warfare Ctr. Carderock Div., West Bethesda, MD)

Loosely coupled schemes for structural-acoustic coupling are examined that obtain the same order of accuracy as the monolithic scheme. The coupling algorithms are implemented in Sierra-SD, a massively parallel finite element application for structural dynamics and acoustics. By adapting the predictor-corrector scheme of Farhat *et al.* (2006), second order time accuracy is achieved with the loosely coupled approach. Node-to-face mappings allow arbitrary discretizations of the structural-acoustic interface. Convergence rates are verified with a one dimensional piston problem with known solution. Numerical results are compared to a shock induced plate experimental benchmark. Computational times for loose and strong coupling are compared for a sphere scattering problem. 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. Department of Energy's National Nuclear Security Administration under contract DE-NA-0003525.

10:25–10:40 Break

10:40

1aPA6. The Iterative Nonlinear Contrast Source method: Basics and extensions. Martin D. Verweij and Elango Selvam (Acoust. Wavefield Imaging, Delft Univ. of Technol., Lorentzweg 1, Delft 2628CJ, Netherlands, m.d.verweij@tudelft.nl)

There are many applications in which the properties of nonlinear waves play a role. For example, in medical diagnostic imaging, the higher harmonics in nonlinear ultrasound are used to reduce clutter from nearby artifacts and to improve resolution, and in non-destructive testing nonlinear mixing products from crossing ultrasound beams are used to assess the state of materials. Simulation of nonlinear waves is important for optimizing equipment and quantitatively assessing phenomena. Unfortunately, analytical approaches cannot deal with complex realistic situations, while traditional Finite Element and Finite Difference methods require extremely large grids to capture waves with many higher harmonics in large three-dimensional domains. The Iterative Nonlinear Contrast Source method was developed to overcome these issues. It is based on an integral equation involving the analytic Green's function of the linearized medium and a distributed contrast source representing nonlinearity. The integral equation is solved iteratively, and the computations are based on Fast Fourier Transforms that only require two grid points of the shortest wavelength. In this presentation, the basic principles behind the method will be explained and its extension to lossy and inhomogeneous media will be shown. Moreover, a novel extension will be presented that enables the computation of nonlinear elastic waves.

Contributed Papers

11:00

1aPA7. Efficient, wide-band rigid-body and elastic scattering computations using transient equivalent sources. John B. Fahnline (ARL / Penn State, P.O. Box 30, State College, PA 16804-0030, jbf103@arl.psu.edu)

Previous analyses by the author have shown that transient structural-acoustic problems can be solved using time stepping procedures with the structure and fluid modeled using finite elements and equivalent sources, respectively. Here, the analysis is extended to include scattering problems. Although scattering problems have been discussed extensively in the literature, the current formulation is unique because it approximates the acoustic coupling matrix as sparse. Also, most of the previous analyses have assumed the problem is time harmonic, and there is an advantage to performing the computations in the time domain because only a limited number of time steps are required to obtain wideband frequency resolution. This is especially true if the main emphasis is on the mid- to high-frequencies since the ringing response is typically dominated by the lowest frequency modes. Several examples are solved to validate the computations and to document the computation times and solution accuracy.

11:15

1aPA8. Radiation force on spheres in standing waves in ideal fluids: Improved simple approximation for material dependence. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

When spheres are much smaller than the acoustic wavelength but sufficiently large that viscous dissipative properties are insignificant, the Yosioka-Kawasima approximation for the radiation force in standing waves is widely used. One consequence is that the force is taken to be proportional to the sphere's volume. One approach to finding how corrections to that approximation depend on material properties is to start with Hasegawa's full partial-wave-based series and convert it to partial-wave phase shifts using analytical relations [L. Zhang and P. L. Marston, J. Acoust. Soc. Am. **140**, EL178–EL183 (2016)]. When all the relevant phase shifts are real and sufficiently small in magnitude, a simple approximation can be found for (possibly imaginary) frequencies giving vanishing forces. That was an unanticipated application. (Material dissipation and surface tension are taken to be insignificant.) Using that frequency root, a modification of the Yosioka-

Kawasima radiation force is derived that allows for the liquid or solid sphere to be finite in size [P. L. Marston, J. Acoust. Soc. Am. **142**, 1167–1170 (2017) and at press]. The force is no longer strictly proportional to the volume. This approach is supported by numerical comparisons with Hasegawa's series when the phase shifts are small. [Research supported by ONR.]

11:30

1aPA9. Geometric approximation of radiation stress projections for canonical objects. Philip L. Marston and Timothy D. Daniel (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

Modulated ultrasonic radiation pressure facilitates the selective excitation of modes of objects through modulation of appropriate surface stress projections [P. L. Marston and R. E. Apfel, J. Acoust. Soc. Am. **67**, 27–37 (1980)]. For some large objects, it can be impractical to evaluate the needed stress projections using standard analytical or computational methods. For highly reflecting objects geometric approximations may provide useful insight into the relevant stress projections. The simplest projection gives the translational radiation force on the object. In that case, ray analysis of the far-field scattering was shown to recover the radiation force on large perfectly reflecting spheres in plane waves [P. L. Marston, J. Acoust. Soc. Am. **120**, 3518–3524 (2006)]. In the present research, the appropriate force limit is also recovered through integrating a geometrically approximated local stress projection on the object's surface. Appropriate limiting forces for other canonical objects or situations involving illumination by beams are also found to be recovered using geometrical methods. It was helpful to generalize some prior geometrical results [P. L. Marston, J. Acoust. Soc. Am. **121**, 753–758 (2007)] for scattering by perfectly reflecting spheres. [Work supported by ONR.]

11:45

1aPA10. Time domain simulation of porous surfaces for comparison with the ANSI impedance measurement. Junjian Zhang and Z. C. Zheng (Aerosp. Eng., Univ. of Kansas, 1530 15th St., Lawrence, KS 66045, JJzhang@ku.edu)

This study is to investigate sound propagation with different porous ground conditions. Sound propagations are simulated using finite difference time domain methods (FDTD). The ground materials are modeled as porous media. Two types of models are used in simulating the porous ground: (1) the original Zwikker-Kosten (ZK) model coupled with the immersed boundary method and (2) a modified ZK impedance model, as proposed by Wilson *et al.* (2006, 2007), as a time-domain boundary condition (TDBC) derived from the frequency-domain impedance boundary condition. The first method directly calculates the wave propagation inside porous media, while the second method only specifies the ground surface as a porous boundary condition. Both methods are based on the ZK-type models but have different responses in the frequency domain. Sound pressure levels in different frequency ranges are studied and discussed. Simulation results from both of the models are compared with the measurement data in ANSI S1.18 standard.

MONDAY MORNING, 7 MAY 2018 — NICOLLET D2, 8:00 A.M. TO 12:00 NOON

Session 1aPP

Psychological and Physiological Acoustics: Consequences of Asymmetrical Hearing

Matthew Goupell, Cochair
Hearing and Speech Sciences, University of Maryland, College Park, 0119E Lefrak Hall, College Park, MD 20742

Lina Reiss, Cochair
Otolaryngology, Oregon Health & Science University, 3181 SW Sam Jackson Park Road, Mailcode NRC04, Portland, OR 97239

Chair's Introduction—8:00

Invited Papers

8:05

1aPP1. Electrode length, placement, and frequency allocation distort place coding for bilateral, bimodal, and single-sided deafened cochlear implant users. David M. Landsberger (Otolaryngol., New York Univ. School of Medicine, 550 First Ave., New York, NY 10016, David.Landsberger@nyumc.org)

There is typically a mismatch between the frequency expected at a given cochlear angle and the frequency presented at that angle by a cochlear implant. The frequency mismatch may be particularly important for the population of cochlear implant users who are combining electric hearing across ears or who have electric hearing in one ear and acoustic hearing in the contralateral ear. If the place coding provided by a cochlear implant is misaligned with place coding in the contralateral ear, then the listener may be provided with inconsistent or even conflicting auditory cues. Due to plasticity, the auditory system may be able to compensate for these misalignments. However, data suggest that the ability of some listeners to fully adapt to the distorted or conflicting place maps is limited. The frequency mismatch from a cochlear implant can be adjusted by changing the electrode length, insertion depth, and frequency allocation. Additionally, changing these variables effect the relationship between a change in input frequency and the corresponding change in place pitch.

8:25

1aPP2. Measuring spectral asymmetry for cochlear-implant listeners with single-sided deafness. Joshua G. Bernstein, Olga A. Stakhovskaya, Kenneth K. Jensen (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 Matthew Goupell (Dept. of Hearing and Speech Sci., Univ. of Maryland, College Park, MD)

Cochlear implants (CI) for individuals with single-sided deafness (SSD) are generally programmed with a standard frequency-allocation table (FAT) to maximize the transmission of speech information through the CI. However, a standard FAT is likely to produce mismatch in interaural place of stimulation, which could limit the spatial-hearing benefits that SSD-CIs provide. This study compared three methods for estimating interaural mismatch for SSD-CI listeners: interaural-time-difference (ITD) discrimination, pitch discrimination, and computed-tomography (CT) scan estimates of electrode intracochlear position. For all three measures, the estimated matched acoustic frequency increased for subsequently more basal electrodes. The ITD-based and CT-based estimates were generally consistent with one another, and were on average about one octave above the standard FAT. The pitch-based estimates were considerably lower, averaging about a half-octave above the standard FAT. These results suggest that pitch-discrimination measurements may be more susceptible to plasticity than ITD-discrimination measurements, and that CT images align with binaural function. The measurements will provide a basis to determine whether reducing mismatch can enhance spatial hearing for SSD-CI listeners. [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.]

8:45

1aPP3. Results in cochlear implant recipients with substantial contralateral acoustic hearing. Jill Firszt (Otolaryngol., Washington Univ. School of Medicine, 660 South Euclid Ave., Campus Box 8115, St. Louis, MO 63110, firsztj@wustl.edu), Ruth Reeder (Otolaryngol., Washington Univ. School of Medicine, Hillsboro, Oregon), Laura Holden, and Noel Dwyer (Otolaryngol., Washington Univ. School of Medicine, St. Louis, MO)

The present study evaluated longitudinal performance outcomes in adult cochlear implant recipients with substantial hearing in the contralateral ear (mean better-ear PTA of 23.7 dB HL; range 0–40 dB HL). Of the 19 participants, 10 used a better-ear (BE) hearing aid both pre- and post-implant. Test intervals included pre-implant, 6 and 12 months post-implant. Participants' everyday listening condition was primarily a single hearing ear pre-implant and bimodal (CI + BE) post-implant. Results indicated significant improvement over time in the everyday listening condition for sentences at a soft presentation level, sentences in noise (either surrounding or toward the BE), and localization. Self-reported communication ability also improved significantly over time. Group mean CNC word scores for the implanted ear alone improved significantly pre- to post-implant. Ear conditions were compared at the 6-month post-implant test interval. The same measures that demonstrated improvement over time in the everyday listening condition demonstrated significant bimodal improvement compared to the BE alone at the 6-month test interval. Bimodal benefit for these participants varied in degree and by measure compared to adult CI recipients with greater HL in the contralateral ear. Cochlear implantation was an effective treatment for individuals relying on a single hearing ear pre-implant.

9:05

1aPP4. Bimodal neglect: When electrically- and acoustically stimulated ears do not play nice with each other. Mario Svirsky and Arlene C. Neuman (New York Univ., 550 First Ave., NBV-5E5, New York, NY 10010, mario.svirsky@nyumc.org)

This study describes speech perception in quiet, before and after implantation, in 54 adult cochlear implant (CI) patients with usable hearing in the ear contralateral to the CI. Patients had preoperative aided CNC word scores of at least 8% as well as stable unaided hearing in the unimplanted ear throughout the analysis period. Word identification scores in quiet were analyzed for the implanted ear only (CI), acoustic hearing ear only (HA), and the bimodal condition (CI + HA) to determine changes in performance over time and to characterize bimodal benefit (difference between bimodal score and best single-ear score). On average, patients experienced 6% bimodal benefit in quiet after implantation. Not surprisingly, speech scores in the implanted ear and in the binaural condition significantly improved post-implantation. However, scores in the HA decreased unexpectedly after implantation in almost one third of the patients despite unchanged PTA. Although simple explanations like hearing aid malfunction or improper fitting cannot be entirely ruled out, it is possible that these decreases in speech perception may be due to centrally driven neglect of auditory input from one ear when the contralateral ear becomes dominant due to highly asymmetric input.

9:25

1aPP5. Difficulty with understanding speech in background noise is predicted by broad binaural pitch fusion in bimodal cochlear implant users. Yonghee Oh (Otolaryngology-Head and Neck Surgery, Oregon Health & Sci. Univ., Mailcode NRC04, 3181 SW Sam Jackson Park Rd., Portland, OR 97239, oyo@ohsu.edu), Nirmal Kumar Srinivasan (Speech-Lang. Pathol. & Deaf Studies, Towson Univ., Towson, MD), Anna C. Diedesch (Commun. Sci. and Disord., Western Washington Univ., Bellingham, WA), Curtis Hartling (Otolaryngology-Head and Neck Surgery, Oregon Health & Sci. Univ., Portland, OR), Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, Portland, OR), and Lina Reiss (Otolaryngology-Head and Neck Surgery, Oregon Health & Sci. Univ., Portland, OR)

Combined use of a cochlear implant (CI) and a contralateral hearing aid has been shown to improve CI users' speech perception performance in multi-talker environments, presumably due to the use of acoustic temporal fine structure cues for separating one talker from several other talkers. However, there is large variability in this bimodal benefit. In this study, we show that differences in width of binaural pitch fusion, the fusion of dichotically presented tones that evoke different pitches across ears, explain a large part of this variability. Specifically, broad binaural pitch fusion could lead to fusion of multiple voices as one voice, and reduce the ability to use voice pitch differences to separate voices and understand speech in a multi-talker environment. Speech reception thresholds measured using male and female target talkers were compared with binaural pitch fusion results. Overall performance improved with different genders for target and maskers. A strong negative correlation was observed between voice gender benefit and breadth of binaural pitch fusion. These results suggest that sharp binaural pitch fusion is necessary for maximal speech perception in noise when acoustic hearing is available to transmit voice pitch cues. [Work supported by NIH-NIDCD Grant Nos. R01 DC01337 and F32 DC016193.]

9:45

1aPP6. Dichotic listening performance and listening effort for asymmetrical inputs. Kristina D. Milvae, Stefanie E. Kuchinsky (Hearing and Speech Sci., Univ. of Maryland College Park, LeFrak Hall, 7251 Preinkert Dr., College Park, MD 20742, kmilvae@umd.edu), Olga A. Stakhovskaya (Hearing and Speech Sci., Univ. of Maryland College Park, Bethesda, MD), and Matthew Goupell (Hearing and Speech Sci., Univ. of Maryland College Park, College Park, MD)

Binaural hearing typically improves speech understanding in difficult listening situations. Cochlear-implant (CI) recipients seek to gain binaural hearing benefits through bilateral implantation. However, the standard of care is currently sequential implantation, possibly with a long difference in the duration of deafness between the ears. Sequential implantation may cause interaural asymmetry in the quality of the signal, which could limit binaural benefits. A range of bilateral CI listeners, from those who experience binaural benefits to those who demonstrate interference, were recruited for this experiment. Listeners attended to one ear and reported digits heard in monotic and dichotic conditions. Perceptual performance and listening effort (via pupillometry) were measured. A control group of normal-hearing listeners were tested and were presented unprocessed, four-, and eight-channel vocoded digits, and the processing was independent across ears. It was hypothesized that attending to a poorer-quality signal in the presence of a better-quality signal would lead to poorer performance than in the monotic condition. Pupillometry was hypothesized to reflect the listener's ability to perceive the poorer-quality signal in the presence of a better-quality signal. Results will inform strategies to maximize binaural hearing in bilateral CI listeners.

10:05–10:20 Break

10:20

1aPP7. Early temporary asymmetrical hearing impairs behavioral and neural sensitivity to sound location. Kelsey L. Anbuhl and Daniel J. Tollin (Dept. of Physiol. & Biophys., Univ. of Colorado School of Medicine, University of Colorado Anschutz Medical Campus, 12631 E 17th Ave., Aurora, CO 80045, Daniel.Tollin@ucdenver.edu)

Children experiencing asymmetrical hearing early in life often display binaural hearing impairments that persist long after symmetric hearing is restored, suggesting abnormal central auditory development. Here, we test the hypothesis that abnormal neural coding of the interaural level difference (ILD) cue to sound location, resulting from asymmetrical hearing during development, underlies the hearing impairments. Guinea pigs were reared for 8 weeks from birth with a unilateral conductive hearing loss (CHL via earplug). After earplug removal we measured behavioral spatial acuity, the binaural interaction component (BIC) of the auditory brainstem response (ABR), and sensitivity to ILDs in auditory midbrain neurons (inferior colliculus, IC). Animals raised with asymmetric hearing displayed worse spatial acuity (~2x worse) for high-pass noise compared to age-matched controls, suggesting impaired sensitivity to ILDs. Physiologically, animals displayed abnormal BICs of the ABR indicating altered binaural processing in the auditory brainstem. Additionally, ILD discrimination thresholds for single neurons in the IC, determined using Fisher Information, were ~2 times worse than that found in controls. These results suggest that experiencing even temporary asymmetric hearing during early development persistently alters the normal development of spatial hearing, which may be attributed to impaired binaural processing at the level of the brainstem and midbrain.

10:40

1aPP8. Auditory and non-auditory effects of asymmetrical hearing loss in animal models. Amanda Lauer (Otolaryngology-HNS, Johns Hopkins Univ. School of Medicine, 515 Traylor, 720 Rutland Ave., Baltimore, MD 21205, alauer2@jhmi.edu)

Animal models offer an opportunity to investigate the consequences of asymmetrical hearing loss with a level of control over extraneous variables that is not possible with human listeners. Experimental manipulations to induce unilateral conductive or sensorineural hearing loss in animal models have demonstrated substantial and diverse changes in physiological function, anatomical reorganization of neural projections, and adverse effects on auditory behaviors. Our experiments in rodents exposed to acoustic trauma with one ear plugged result in asymmetrical hearing loss and abnormal auditory brainstem response waveforms that occur when either the unprotected or protected ear is stimulated. Susceptibility to auditory brainstem response deficits appears to differ between the right and left sides. Asymmetrical hearing loss is associated with increased startle reactivity to loud sounds, suggesting abnormal loudness perception or an increased emotional response to sound. These animals also show abnormal social interaction, and some subjects show reduced exploratory behavior in a novel environment indicative of heightened anxiety. These studies provide a framework for investigating the interplay between the auditory and non-auditory behavioral consequences of hearing loss, as well as the underlying neuropathology.

11:00

1aPP9. Talk to my better ear: Consequences of asymmetric bilateral hearing in development. Karen A. Gordon, Melissa Polonenko, and Blake Papsin (The Hospital for Sick Children, Rm. 6D08, 555 University Ave., Toronto, ON M5g 1x8, Canada, karen.gordon@utoronto.ca)

We study consequences of asymmetric hearing in children with bilateral profound deafness who used a unilateral cochlear implant before bilateral implantation and children who had access to acoustic hearing in one ear with or without a hearing aid before receiving a cochlear implant in the opposite ear (bimodal). Electrophysiological and behavioral measures post-implantation reveal benefits of bilateral implant/bimodal device use but also persistent consequences of early asymmetric auditory input. Auditory cortices maintain an aural preference for the first hearing ear despite chronic bilateral/bimodal hearing. Asymmetries are also evident in speech detection and perception; children receiving bilateral cochlear implants with long delays continue to favour their first implanted ear in both quiet and noise. Children with bimodal devices who have insufficient hearing in the non-implanted ear begin to prefer to listen with the cochlear implant. Impaired binaural hearing in groups with asymmetric function are evidenced by poor integration of input from the two devices and impaired perception and cortical processing of interaural timing differences. Together, these data suggest that children who had a period of asymmetric hearing in early life continue to have aural preference for a "better ear" which could compromise development of binaural/spatial hearing.

11:20

1aPP10. The effects of early acoustic hearing on speech perception, language, and literacy abilities of pediatric cochlear implant recipients. Lisa S. Davidson (Otolaryngology, Washington Univ. School of Medicine in St. Louis, 4523 Clayton Ave., Campus Box 8115, St. Louis, MO 63110, davidsonls@wustl.edu)

The primary aim of our current project is to determine the degree to which a period of hearing aid (HA) use facilitates language development in children following the receipt of their first cochlear implant (CI). We tested 117 pediatric CI recipients ranging in age from 5 to 8 years on tests of speech perception and standardized tests of receptive vocabulary and language. Follow up testing (ages 7–10 years) included early literacy assessments. The speech perception test battery includes word recognition in quiet and noise (segmental perception), talker and stress discrimination, and emotion identification (suprasegmental perception). A continuum of residual hearing levels and length of HA use are represented: some children have bimodal devices, while others received their second CI either simultaneously or sequentially at varying time intervals since their first CI (mean age at 1^{st} CI/s = 2.1 years). Comprehensive threshold and device histories were collected on all CI recipients. Based on our analyses, we identified a critical duration of HA use for each of three ranges of hearing loss severity (pure-tone-averages of 72, 92, and 110 dB HL) that maximizes suprasegmental perception (which operates independently from segmental perception to promote spoken language and literacy).The clinical implications of these results will be discussed.

11:40

1aPP11. Interhemispheric auditory cortical relationships in asymmetric hearing loss. Steven W. Cheung (Otolaryngology-Head and Neck Surgery, Univ. of California San Francisco, 2233 Post St., Ste. 341, San Francisco, CA 94115, Steven.Cheung@ucsf.edu)

Directional acoustic trauma, sudden hearing loss with incomplete recovery, Meniere's disease, and cerebellopontine angle tumors are common clinical conditions that present with asymmetric hearing loss (ASL). Presentation of input signals from both ears for interhemispheric processing may be necessary to achieve the best hearing outcomes. To review temporal and spatial interhemispheric functional relationships revealed by microelectrode mapping studies in an animal model of ASL and by functional imaging studies in human single-sided deafness (SSD). Animal and human cohort comparison studies by investigators at the University of California. In animals with mild-to-moderate ASL, interaural temporal difference is a strong driver of cortical plastic change. Six months after hearing loss induction, the two cortices reorganize into a state of relative local hemispheric autonomy. In humans with SSD, temporal and spatial cortical plastic change is evidenced by reduced interhemispheric mean difference of the M100 peak response latency and increased activation spread distance in the hemisphere contralateral to the only hearing ear and decreased distance in the ipsilateral hemisphere in SSD subjects to tonal stimuli, respectively. Variations in hearing outcomes for acoustical and electrical rehabilitation of the poorer ear in ASL may be related particular states of interhemispheric relationships.

Session 1aSP

Signal Processing in Acoustics, Underwater Acoustics, and Architectural Acoustics: Signal Processing for Complex Environments. I

Sandra L. Collier, Cochair
U.S. Army Research Laboratory, 2800 Powder Mill Rd., RDRL-CIE-S, Adelphi, MD 20783-1197

Kainam T. Wong, Cochair
Dept. of Electronic & Information Engineering, Hong Kong Polytechnic University, 11 Yuk Choi Road, Hung Hom KLN, Hong Kong

Max Denis, Cochair
U.S. Army Research Lab., 2800 Powder Mill Road, Adelphi, MD 20783-1197

Invited Papers

9:00

1aSP1. Outdoor acoustic tomography using retrieved Green's function from controlled-sources: An Experimental Feasibility Study. Max Denis (U.S. Army Res. Lab., 2800 Powder Mill Rd., RDRL-CIE-S, 1 University Ave., Lowell, Massachusetts 01854, max_f_denis@hotmail.com), Sandra L. Collier, John Noble, W. C. Kirkpatrick Alberts, David Ligon, Leng Sim, Christian G. Reiff, and Deryck D. James (U.S. Army Res. Lab., 2800 Powder Mill Rd., RDRL-CIE-S, Adelphi, MD)

In this work, outdoor acoustic tomography is investigated using the retrieve Green's function originating from a virtual source located inside a region of interest (ROI). Of particular interest are the location and amplitude of reflectors, as well as the extraction of atmospheric conditions. Existing methods utilize Green's function retrieval approaches, such as cross-correlation (CC) and multidimensional deconvolution (MDD), of reflection data from inhomogeneities (virtual sources) within the ROI. This allows one to retrieve the Green's function without detailed knowledge of the medium itself. The imaging resolution, artifacts, and close match with the true reflectors are investigated and discussed for both Green's function retrieval methods.

9:20

1aSP2. Propagation effects on acoustic particle velocity sensing. Sandra L. Collier, Max Denis, David Ligon, Latasha Solomon (U.S. Army Res. Lab., 2800 Powder Mill Rd., RDRL-CIE-S, Adelphi, MD 20783-1197, sandra.l.collier4.civ@mail.mil), Madeline M. Erikson (U.S. Military Acad., West Point, NY), John Noble, W. C. Kirkpatrick Alberts, Leng Sim, Christian G. Reiff, and Deryck D. James (U.S. Army Res. Lab., Adelphi, MD)

The effects of atmospheric turbulence on the acoustic particle velocity are examined for both narrow-band and wide-band sources. The statistical distributions of the acoustic particle velocity and pressure fields are analyzed for a series of field tests. Applications to signal processing with particle velocity sensors are considered.

9:40

1aSP3. Obtaining acoustic intensity from multisource statistically optimized near-field acoustical holography. Trevor A. Stout (Brigham Young Univ., N283 Eyring Sci. Ctr., Provo, UT 84602, tastout6@gmail.com), Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH), Kent L. Gee, and Tracianne B. Neilsen (Brigham Young Univ., Provo, UT)

Multisource statistically optimized near-field acoustical holography (M-SONAH) improves the field reconstruction process by directly incorporating into the pressure propagator types of wavefunctions that correspond most closely to the source geometries of interest [A. T. Wall *et al.*, J. Acoust. Soc. Am. **137**, 963–975 (2015)]. The M-SONAH method has previously been used to localize acoustic sources in a full-scale jet engine plume above a rigid reflecting plane by adding a second set of cylindrical wavefunctions corresponding to the image source [A. T. Wall *et al.* J. Acoust. Soc. Am. **139**, 1938–1950 (2016)]. Here, M-SONAH theory is extended to obtain the vector particle velocity and, by extension, the acoustic intensity. Discussed are two examples that relate to the full-scale jet noise-with-image-plane reconstruction problem: (1) a Gaussian line source with image and (2) a jet-like wavepacket and image, with hologram geometry identical to that of the full-scale experiment. The results from both examples reveal intensity errors less than 3 dB and 10 degrees within the top 20 dB of the reconstruction region. The results also suggest that intensity reconstruction magnitudes less than those obtained at the measurement aperture edges should be discarded. [Work supported by ONR.]

10:00

1aSP4. Directional pointing error in "Spatial Matched Filter" beamforming at a tri-axial velocity-sensor with non-orthogonal axes. Chibuzo J. Nnonyelu (Electron. and Information Eng., The Hong Kong Polytechnic Univ., BC606 (Table 3), 11 Yuk Choi Rd., Hung Hom, Hung Hom, Kowloon 999903, Hong Kong, joseph.nnonyelu@connect.polyu.hk), Charles H. Lee (Dept. of Mathematics, California State Univ., Fullerton, CA), and Kainam T. Wong (Electron. and Information Eng., The Hong Kong Polytechnic Univ., Hung Hom, Hong Kong)

The "tri-axial velocity-sensor" has three axes that are nominally perpendicular, but may be non-perpendicular in practice, due to real-world imperfections in manufacturing, deployment, or maintenance. This paper investigates how such non-perpendicularity would affect the tri-axial velocity-sensor's azimuth-elevation beam-pattern in terms of the beam's pointing direction.

10:20–10:35 Break

10:35

1aSP5. Sparse array performance in channels of limited coherence. Andrew T. Pyzdek (Graduate Program in Acoust., Penn State Univ., PO Box 30, State College, PA 16804, atp5120@psu.edu) and David C. Swanson (Appl. Res. Lab., The Penn State Univ., State College, PA)

Sparse arrays reduce the number of redundant measurements for lags on an array in comparison to fully populated arrays. For ideal measurements, where spatial stationarity is maintained and measurement error is minimal, the performance of such arrays can be shown to be similar to that of a uniform array of the same length with appropriate processing techniques. However, for complex environments with limited spatial coherence or noisy measurements, array sparsity can impact performance. We construct a simple model that relates the relative positions of redundant measurements and channel coherence length to the variance of covariance function estimates, determine a set of lag measurement weightings that minimize said variance for a given set of redundant lag positions, determine a set of conditions on which a uniform weighting is optimal, and demonstrate that regularly spaced lag repetitions optimally sample a spatially varying covariance function.

10:55

1aSP6. Evaluating the effectiveness of subarrays in beamforming computations of complex jet noise source environments. David F. Van Komen, Tracianne B. Neilsen, Blaine M. Harker, Kent L. Gee, S. Hales Swift (Phys. and Astronomy, Brigham Young Univ., N283 Eyring Sci. Ctr., Provo, UT 84602, david.vankomen@gmail.com), Alan T. Wall (Air Force Res. Lab., Wright-Patterson AFB, OH), Micah Downing, and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Conventional acoustic beamforming methods, while powerful for localizing acoustic sources, require an uncorrelated monopole assumption, depend on the input array location, and are unreliable in a complex source environment, such as jet engine noise. The complexities arise from the large, partially correlated, extended source region, the multiple types of noise sources, such as directional and omnidirectional, and partially correlated sources that violate the integral uncorrelated monopole assumption of beamforming. The complexity of these sources requires advanced beamforming methods. The aim of this research is to determine what advanced beamforming methods tell us about complex, full-scale jet noise sources. To learn about these sources, an array spanning the entire length of a jet from an F-35 aircraft is split into multiple subarrays. Application of the beamforming algorithm to a subarray in the maximum sound region localizes the directional sources present in jet noise; a subarray to the side of the engine characterizes more omnidirectional sources; and a subarray farther forward yields the approximate origin of broadband shock-associated noise. Using these advanced beamforming methods, combined with the subarray processing, opens new insights in this complex environment. [Work supported by the Air Force Research Laboratory; data courtesy of F-35 JPO.]

11:15

1aSP7. The blended dominant mode rejection adaptive beamformer. John R. Buck (ECE Dept., UMass Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, jbuck@umassd.edu), Andrew Singer (ECE Dept., Univ. of Illinois Urbana-Champaign, Champaign, IL), and Kathleen E. Wage (ECE Dept, George Mason Univ., Fairfax, VA)

Adaptive beamformers balance competing demands of enhancing desired signals, suppressing discrete interferers and attenuating background noise. This requires an accurate estimate of the spatial covariance matrix that is challenging to obtain in nonstationary environments. The dominant mode rejection (DMR) beamformer [Abraham & Owsley, Oceans, 1990] addresses this challenge by replacing the sample covariance matrix (SCM) eigenvalues of the noise subspace with the average of these eigenvalues to improve its conditioning. The DMR beamformer requires an estimate of the interferer subspace dimension, and the DMR beamformer's performance degrades when this estimated dimension is inaccurate. We propose the Blended DMR beamformer, whose array weights are formed as a mixture of fixed-dimension DMR beamformer array weights across a range of interferer dimensions. The contribution of each fixed-dimension DMR beamformer is a function of its recent performance in suppressing interferers and attenuating noise. The performance of this blended DMR beamformer approaches that of the best fixed-dimension DMR beamformer in stationary environments, and can outperform any fixed DMR beamformer in many nonstationary environments. [Work supported by ONR Code 321US.]

11:35

1aSP8. Frequency-difference beamforming with multiple realizations in an inhomogeneous environment. Alexander S. Douglass and David R. Dowling (Mech. Eng., Univ. of Michigan, 2010 AL, 1231 Beal, Ann Arbor, MI 48109, asdougl@umich.edu)

Frequency-difference beamforming is a sparse-array signal processing method that shifts the signal analysis to lower, out-of-band (difference) frequencies, provided that the difference frequency does not exceed the signal bandwidth. In an inhomogeneous environment with discrete scatterers, low-frequency fields may not be significantly affected, provided the scatterers are small relative to the

signal wavelength. However, when the scatterer sizes are comparable to the signal wavelength, scattering effects are likely to be more significant. Previously, it was shown that frequency-difference beamforming can overcome some of the performance degradation from the high frequency scattering by shifting the processing to lower, out-of-band frequencies. In this case, the frequency-difference output was comparable to the output using a lower-frequency signal in the same environment. In this talk, theoretical fields that include scattering at the in-band frequencies are considered to predict the performance of the frequency-difference method. Furthermore, simulations and experiments in a 1.07-m-diameter water tank with a sparse array are used to determine the performance and robustness of the method in the presence of discrete high-contrast scatterers. Multiple realizations of randomly placed scatterers are combined to quantify performance statistics. [Sponsored by NAVSEA through the NEEC, and by ONR.]

MONDAY AFTERNOON, 7 MAY 2018 NICOLLET C, 1:00 P.M. TO 4:15 P.M.

Session 1pAA

Architectural Acoustics and Noise: Architectural Acoustics Experimental Methods in Laboratories

Jin Yong Jeon, Cochair
Department of Architectural Engineering, Hanyang University, Seoul 133-791, South Korea

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

Invited Papers

1:00

1pAA1. Portable goniometers with high angular resolutions and large radii for experimental investigations of acoustic diffractions. Jonathan Kawasaki, Ning Xiang, Jolene Stoffle (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, kawasj@rpi.edu), and Peter D'Antonio (RPG Acoust. Systems, LLC, Passaic, NJ)

Surface scattering and diffuse reflections of acoustic diffusers are of relevance in room-acoustics simulations or for characterization purposes. Polar responses of diffraction objects to acoustic plane-wave excitations can be experimentally characterized using a circular microphone array: acoustical goniometer. Recently, the Graduate Program in Architectural Acoustics at Rensselaer Polytechnic Institute has begun to explore the advanced physical theory of diffraction (PTD) [Xiang and Rozynova, *J. Acoust. Soc. Am.*, **141**, pp. 3785 (2017)]. In order to validate the PTD applied to a number of canonical shapes (rigid wedges or plates) of finite sizes, experimental investigations on the diffraction objects need to be carried out with high angular resolutions and sufficiently wide angular ranges. A large radius of the acoustical goniometer is also crucial to satisfy acoustical far-field conditions. This talk will discuss the effort in implementation and processing algorithms for various radii of portable acoustical goniometers between 1 m and 4 m with angular resolutions up to 1.25 degrees, that can be easily deployed in indoor empty spaces of sufficient dimensions for an angular range across 180 degrees and beyond when diffusers or diffraction objects with one-axis symmetry need to be experimentally characterized.

1:20

1pAA2. High spatial resolution scanning for experimental room-acoustic measurements in scale models. Daniel Tay, Ning Xiang, and Aditya Alamuru (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., School of Architecture Rensselaer Polytechnic Inst. 110 8th St. - Greene Bldg., Troy, NY 12180, tayd@rpi.edu)

This paper presents an experimental study of low frequency behavior of sound propagation in acoustically coupled volumes. The experiments are carried out in an eighth-scale model of two coupled volumes with an automated high spatial-resolution scanning system. Both sound pressure and three-dimensional sound intensities are automatically measured in terms of room impulse responses at each scanning positions over densely defined two-dimensional grids covering the entire areas of the two coupled rooms that enable the analysis and visualization of sound energy and sound energy flows. Through these measurements, and with the aid of a predictive computational model, this paper will discuss low frequency modal behaviors of sound energy flows within one of two coupled volumes. As such, the distribution of potential and kinetic energies, as well as that of active and reactive sound intensities within the coupled space will be separately analyzed. The distributions of intensity vector fields are compared through computational simulation. Consequently, a discussion on possible discrepancies in modal distributions between the computational and experimental results rounds off this paper.

1:40

1pAA3. The irrelevant speech effect in multi-talker environments: Applications to open-plan offices. Manuj Yadav, Densil Cabrera (The Univ. of Sydney, Rm. 469, Wilkinson Bldg., 140, City Rd., Darlington, NSW 2008, Australia, manuj.yadav@sydney.edu.au), Lucas Brooker (Sydney, NSW, Australia), James Love, Jungsoo Kim, and Richard de Dear (The Univ. of Sydney, Sydney, NSW, Australia)

Irrelevant speech from co-workers has consistently been listed as one of the major nuisances within open-plan work environments. To explore the psychoacoustic basis of the so-called irrelevant speech effect (ISE; which causes cognitive decline and increases acoustic distraction) within a multi-talker context, a series of laboratory-based experiments were run. These experiments used room acoustical simulations of various open-plan office environments within a climate-controlled chamber to test several factors relevant to the ISE. These factors included the number of simultaneously active talkers, their spatial arrangements, sound pressure levels and the semanticity of speech. The main findings thus far indicate that multi-talker speech lead to a stronger ISE than speech from a single talker, which further interacts with the overall sound level, semanticity, etc., to exhibit a more complicated effect than what has been proposed in previous studies. The effect of the number of active talkers alone suggests reconsiderations of some assumptions in the ISO 3382-3. More generally, the perception of speech "babble" seems to be affected by multiple factors in a realistic working environment. Some of our research is directly related to characterising speech babble and its function as a beneficial sound masker in multi-talker environments.

2:00

1pAA4. Experiments with high-back chairs and retroreflective surfaces for increasing the support of one's own voice over short conversational distances. Densil Cabrera, Manuj Yadav, Dagmar Reinhardt, Jonothan Holmes, Beau Ciccarello, Hugo Caldwell, and Adam Hannouch (The Univ. of Sydney, Rm. 589, Wilkinson Bldg., 140, City Rd., NSW 2006, Australia, manuj.yadav@sydney.edu.au)

Acoustic support of one's own voice affects speech, and increased support can foster more relaxed voice projection, which reverses the Lombard effect. While hearing one's own voice in typical rooms shows subtle influences of "global" room acoustics, local treatment can yield stronger effects for talking-listeners. This paper considers two types of architectural acoustic treatment for supporting one's own voice and modifying speech propagation—high-back chairs and retroreflective ceilings. For a talking-listener, local acoustic treatment such as high-back chairs can be designed to selectively attenuate ambient noise while providing enhanced reception of sound from a particular direction, project speech towards a listener with increased gain for speech intelligibility, and also provide voice support. Acoustically retroreflective surfaces (e.g., ceilings and vertical partitions) provide increased voice support by reflecting a person's voice back to them, without such local treatment. This also has the advantage of reducing the transmission of speech beyond the conversation, providing a way to ameliorate speech distraction in multi-talker environments. This paper describes experiments in which the voice support of such treatments is quantified, via oral-binaural transfer functions. Results are supported with numerical simulation (finite-difference time-domain), and discussed in relation to practical design solutions for speech privacy and vocal comfort.

2:20

1pAA5. Diffusion: How new research and data formats can be of use in simulations. Ronald Sauro (NWAA Labs, Inc., 90 Tower Blvd., Elma, WA 98541, audio_ron@msn.com)

We will explore the new ways of measuring diffusion using free field methods and how those resulting data formats can be used in simulation programs such as EASE etc. This includes the calculation of complex balloons that reflect the total directivity of entire arrays of both diffractive and geometric styles of "diffusors." We will show actual measurements vs simulations. Included will be research done by Ron Sauro and extensions done by Jim DeGrandis that include some new graphs that extend our usage of phase data in an understanding of the psychoacoustic behavior of the human brain and our perception of how much data is of use to our designs.

2:40–2:55 Break

2:55

1pAA6. Old problems, new solutions: Architectural acoustics in flux. Ronald Sauro (NWAA Labs, Inc., 90 Tower Blvd., Elma, WA 98541, audio_ron@msn.com)

Many problems that have plagued architectural acoustics for years and have escaped real solutions because of a lack of research and measuring facilities. We will explore the types of problems that are still around and the facilities that are needed to explore those problems. This will include reverberation rooms that have the abilities to explore VLF problems in absorption and transmission loss as well as sound power problems at these irritating very low frequencies. This exploration will include looking at present day abilities including specifications of the future rooms that are needed. Extended frequency response information is needed in today's litigious society when everything offends someone else and helps consultants provide solutions to present and future problems.

Contributed Papers

3:15

1pAA7. Field testing of in-room impact noise. John LoVerde and David W. Dong (Veneklasen Assoc., 1711 16th St, Santa Monica, CA 90404, jloverde@veneklasen.com)

The sound generated by footfall within the source space, sometimes called in-room impact noise, affects the noise level due to movement within a room and also the occupants' perception of quality. Initial studies are based on the standard tapping machine as the source. While no ASTM standard yet exists for measuring the source room sound level, source room measurements with the tapping machine have been performed on several recent laboratory testing programs. The authors have measured source room power levels in the field using the same flooring products that were measured in the laboratory. The data are analyzed to evaluate the correlation between the laboratory data and the resultant noise level realized in the field. The usefulness and applicability of future laboratory and field in-room impact noise standards is discussed.

3:30

1pAA8. Impact noise of bare concrete slabs. David W. Dong, John LoVerde (Veneklasen Assoc., 1711 16th St, Santa Monica, CA 90404, jloverde@veneklasen.com), and Michel Morin (MJM Acoust. Consultants Inc, Montreal, QC, Canada)

A poured concrete slab with no finish flooring and no ceiling is one of the simplest floor assemblies possible. Most structural slabs share similar design parameters such as density, mix, and compressive strength, and might be expected to have a similarly limited range of impact noise isolation. The authors have compiled a large number of bare slab tapping tests performed in laboratories, which provides information on the uncertainty in the test method for the specific assemblies used in the laboratory tests. The authors have compiled a large number of field impact tests on bare slabs in buildings, and the data reveals a surprisingly large variation in the impact sound pressure levels for actual slabs. The effects of this variation is discussed, including the suitability of various ratings of impact sound insulation, the relationship between laboratory and field measurements, and the uncertainty of impact noise measurements.

3:45

1pAA9. Drop Tower Testing—The next step in heavy weight impact testing. Matthew V. Golden (Pliteq, 616 4th St., NE, Washington, DC 20002, mgolden@pliteq.com)

Over the last few years, research into heavy hard impacts from fitness activities has become an active field of investigation and product development. In order in facilitate our efforts, Pliteq has built a custom drop tower capable of repeatability dropping up to 150 kg from well over 2 m height. During the impact, the force, displacement, velocity, and acceleration can be obtained. This paper will give an overview of this apparatus, the type of data that it can collect, how this data can be used to make better products and how the data can be used to predict field performance.

4:00

1pAA10. Finite element modeling of Rayleigh wave propagation in a laboratory sand tank. Anthony F. Ragusa, James H. Miller, and Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, 15 Receiving Rd., Narragansett, RI 02882, afragusa@my.uri.edu)

Rayleigh waves were measured in a laboratory sand tank using an array of accelerometers. This tank allows for the measurement of interface waves for various sediment and soil arrangements in a well-controlled laboratory setting. Rayleigh waves were excited by simple impulsive sources including dropped balls and hammer impacts. In this study, a finite element modeling tool developed by Sandia National Laboratories was used to simulate the sand tank during these excitations. Different modeling scenarios were investigated. Time series of the Rayleigh wave arrivals were modeled at the sensor locations using time domain analysis. A calibrated bender element system was also developed to provide direct measurements of the shear wave speed profile at selected depths. These data were used to develop the material property input parameters for the finite element analysis. The simulated time series at the sensor locations are compared to the accelerometer data. [Work supported by Army Research Laboratory and BOEM.]

MONDAY AFTERNOON, 7 MAY 2018 LAKESHORE B, 1:20 P.M. TO 3:50 P.M.

Session 1pAB

Animal Bioacoustics: Bat and Biomimetic Sonar/Marine Mammal Hearing and Sound Production

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

Contributed Papers

1:20

1pAB1. Biosonar beampattern measurements in flying old world leaf-nosed bats. Yanan Zhao, Hui Ma (Shandong University - Virginia Tech Int. Lab., Shandong Univ., Shandong University Central Campus, Jinan, Shandong 2501000, China, 820123652@qq.com), and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Emission beampatterns, i.e., emitted ultrasound amplitude as a function of direction and frequency, could be an important characteristic of a bat's biosonar since they determine which part of an environment is illuminated with a given frequency and at a given time. Numerical results have predicted intricate geometrical features such as multiple peaks and notches in bat biosonar beampatterns. Furthermore, it has been shown that bats with nasal emission change the shapes of their emission baffles, "noseleaves," during pulse emission. The work presented here is aimed at understanding if and how these two aspects (beam geometry and emission dynamics) are integrated into the bats' biosonar behaviors on the wing. The experiments are conducted with Old World leaf-nosed bats (*Hipposideros pratti* and *Hipposideros armiger)* that have a pronounced noseleaf dynamics. The animals fly in a tunnel that is instrumented with synchronized arrays of high-speed video cameras and microphones. The setup allows for precise tracking of the bats with high spatial resolution as well as beampattern estimates as a function of time. A first pilot dataset has indicated the presence of local shape features in the beampattern that can also vary with time. However, further testing is needed to corroborate these findings.

1:35

1pAB2. Characterization of pinna motion patterns in hipposiderid bats. Peiwen Qiu and Rolf Müller (Mech. Eng., Virginia Tech, 1075 Life Sci. Cir, Blacksburg, VA 24061, peiwenq@vt.edu)

Hipposiderid bats have a set of highly differentiated pinna muscles that support different pinna motion patterns during biosonar behaviors. It has been shown that these bats' pinna motions fall into two separate categories: Rigid motions that change only pinna orientation and non-rigid motions that also change pinna shape. It remains unclear if further subdivisions of these two categories (e.g., different orientations for rigid motions and different deformation patterns for non-rigid motions) can be made. In order to investigate this question, experiments have been conducted where the positions of

a dense set of landmark points (50–80 per pinna) were reconstructed based on footage from an array of four high-speed video cameras. The resulting point trajectory data (positions as a function of time) was clustered to facilitate the detection of patterns. Clustering revealed a structure of coherent surface patches for rigid as well as non-rigid motion. For the rigid motion pilot data set, these patches followed a constant pattern that reflects different motion directions. Within the non-rigid motion pilot data, there were similarities as well as pronounced differences among the analyzed motion sequences, which leads us to formulate the hypothesis that there could be more than one type of non-rigid pinna motion.

1:50

1pAB3. System integration for a biomimetic dynamic sonar head. Dean Conte (Mech. Eng., Virginia Tech, 1075 Life Sci. Cir, Blacksburg, VA 24061, dconte97@vt.edu), Lucas Mun, Shubham Dawda, Brandon Walker (Elec. and Comput. Eng., Virginia Tech, Blacksburg, VA), Joseph Sutlive (Translational Biology, Medicine, and Health Program, Virginia Tech, Roanoke, VA), and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Horseshoe bats (Rhinolophidae) have dynamic biosonar systems with interfaces for ultrasonic emission/reception that deform while diffracting the outgoing/incoming sound waves. This peripheral dynamics has been shown to enhance sensory information encoding. To investigate the properties and potential uses of these system features in the real world, a biomimetic sonar head is being developed with the goal of mimicking as much of the bats' peripheral dynamics as faithfully as possible. Further constraints on this development were suitability for integration with mobile platforms and support for interactive quantitative/in-depth experimentation. Flexible noseleaf and pinna shapes along with an actuation system have been designed to mimic the static geometry of the respective structures in bats as well as several of the animals' degrees of freedom in deforming them. The electrical subsystem has been designed to support high-quality ultrasound generation and recording. Generation of all source voltages and signal amplification have been consolidated into a single circuit board to reduce weight and complexity of the overall system. The frame has been altered to reduce weight and add modularity to the noseleaf and pinna elements. The software architecture is based on a backend (Python, LabVIEW) combined with a flexible front-end (MATLAB) for experimental design and data analysis.

2:05

1pAB4. Beamwidth in bat biosonar and man-made sonar. Rolf Müller and Bryan D. Todd (Mech. Eng., Virginia Tech, ICTAS Life Sci. District (Mail Code 0917), Virginia Tech, Blacksburg, VA 24061, rolf.mueller@vt.edu)

Beamwidth measures how a sonar system distributes emitted energy or receiver sensitivity over angle. For man-made sonars, this is considered to be of critical importance, since a small beamwidth supports high-resolution imaging. For bat biosonar, the role of beamwidth is far less clear. In order to investigate how biosonar beamwidths could compare to its man-made peers, we have compiled beamwidth data for bats (based on numerical predictions) and man-made sonar systems (based on data sheets). An analysis of this data shows that bat biosonar and man-made sonars operate in very different regimes with respect to the ratio of characteristic dimension (i.e., aperture diameter for bats and array length for man-made sonars) to wavelength. Whereas the aperture diameters of bats ranged from falling below the employed wavelength to being about 10 times larger, man-made sonars were typically at least one order of magnitude larger than the employed wavelengths and ranged up to 1000 times larger. Furthermore, for any given characteristic-dimension-to-wavelength ratio, bats showed much more variability in beamwidth than man-made sonar. In fact, the variability in the bat beamwidths was found to be as large as that in a set of random horns, which indicates a role for other determining factors.

2:20

1pAB5. The problem of finding passageways in foliage with biomimetic sonar. Ruihao Wang, Xin Jin, Haosen Wang, and Rolf Müller (Mech. Eng., Virginia Tech, 1075 Life Sci. Cir, Blacksburg, VA 24061, ruihaow1@vt.edu)

Bats living in densely vegetated habitats must have the ability to find passageways in foliage. To explore the sensing problem posed by this navigation task, a sonar head that mimicked a bat's biosonar periphery and emitted a bat-like chirp was placed on a linear track to scan artificial foliages (plastic leaves resembling broad-leaf foliage) interrupted by gaps of controllable width. The first approach to detecting the gaps tested has been based on comparing echo energy levels received when facing a gap as opposed to a closed foliage. The ability to detect a gap in this way depends on the employed beamwidth and the distance to the foliage, i.e., the narrower the beam and closer the sonar head, the narrower a gap can be detected. However, narrower beams were also found to increase the variability in the energy of the returned echoes which could impact the ability to reliably distinguish the different levels of echo energy associated with closed foliage and gaps. Since bats tend to have wide beams due to their limited size relative to the employed wavelengths, it may be hypothesized that the animals have evolved alternative approaches to improve the ability of detecting narrow passageways at a distance.

2:35

1pAB6. Assessing continuous sensory information encoding by a biomimetic dynamic sonar emitter. Luhui Yang (Shandong University - Virginia Tech Int. Lab., Shandong Univ., 1075 Life Sci. Cir, Blacksburg, VA 24061, 913022794@qq.com) and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Bat species in the rhinolophid and hipposiderid families can deform the shapes of their noseleaves and pinnae when emitting/receiving ultrasound. Prior work has shown that these deformations enhance the encoding of sensory information. So far, quantifications of the impact on sensory information encoding have been based on small numbers of discrete shapes and clustering of head-related transfer functions into discrete alphabets. In the current work, an improved estimator for mutual information that is based on entropy estimates from k-nearest neighbor distances (Kraskov *et al*., 2004) was used to overcome these limitations. With this approach, mutual information was estimated based on individual head-related transfer functions along the entire duration of biosonar pulses that were represented by high sampling rates suitable for ultrasonic signals. Testing was carried out with a data set from a dynamic sonar emitter inspired by the noseleaf of Pratt's roundleaf bat (*Hipposideros pratti*). The results were found to be in good agreement with the cluster-based approach. The high temporal resolution employed showed that mutual information between beampatterns decreased very quickly within the pulse duration demonstrating that the noseleaf dynamics can provide the bats with the equivalent of several nearly independent views of the environment within a single pulse.

2:50–3:05 Break

3:05

1pAB7. Extensible detection and classification in Ishmael. David K. Mellinger, Curtis Lending, Sharon L. Nieukirk, Sara L. Heimlich (Coop. Inst. for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, David.Mellinger@oregonstate.edu), Julie N. Oswald, and Michael Oswald (Bio-Waves, Inc., Encinitas, CA)

Ishmael is a bioacoustic analysis software package that has offers automatic detection of animal calls via a number of user-configurable detection methods. Ishmael now offers three significant improvements. The first is that users can download pre-configured detectors from within Ishmael for several cetacean species, including several mysticetes and several odontocetes. Each such detector is characterized with performance-measurement curves (receiver operating characteristic, detection error tradeoff, precision-recall) for a test dataset. Information on the signal-to-noise ratio of test datasets is also provided, since performance information is nearly meaningless without some indication of call clarity. The second is that Ishmael is extensible via a MATLAB interface that allows users to write their own detection plug-ins. Ishmael sends sound data and associated metadata to the plug-in, which analyzes the data and sends back detections and classifications for Ishmael to further handle as it would handle detections from built-in detectors. The third is an interface to Real-time Odontocete Call Classification Algorithm (ROCCA): Ishmael detects clicks and whistles of odontocetes, measures various acoustic characteristics, and sends the resulting acoustic parameters to ROCCA. ROCCA classifies both individual vocalizations, and also groups successive vocalizations together into "encounters," which are then classified separately. [Work supported by LMR and ONR.]

3:20

1pAB8. A decade of marine mammal acoustical presence and habitat preference in the Bering Sea. Kerri Seger (School of Marine Sci. and Ocean Eng., Univ. of New Hampshire, 9331 Discovery Way, Apt. C, La Jolla, CA 92037, kseger@ucsd.edu) and Jennifer L. Miksis-Olds (School of Marine Sci. and Ocean Eng., Univ. of New Hampshire, Durham, NC)

As Arctic seas rapidly change with increased ocean temperatures and decreased sea ice extent, traditional marine mammal distributions may be altered, and non-traditional Arctic species may shift poleward. Extant and seasonal continental shelf odontocete species in the Arctic Ocean include sperm whales, killer whales, beluga whales, delphinids, harbour porpoises, and Dall's porpoises. Until recently, recording constraints limited higher sampling rates, preventing detection of many of these high frequency-producing species in the Arctic seas. Using one of the first long-term data sets to record clicks, buzzes, and whistles, multiple species have been detected and classified to explore shifting distributions in the Arctic Corridor. The data were then paired with environmental variables for generalized linear and additive modeling. Pacific white-sided dolphins were detected for a few years during minimal ice coverage. Dall's porpoises and Risso's dolphins may have been acoustically detected in areas farther north than previously documented in acoustical records. Of all the species, beluga whales are the most highly coupled to the ice edge. Adjusted habitat distributions, suggestions for newly present species to be aware of during visual surveys, and environmental model results will be reviewed in this presentation. [This work was funded by the Office of Naval Research.]

3:35

1pAB9. Binaural AEP audiograms in seven Beluga whales (*Delphinapterus leucas*) from the Okhotsk Sea population. Evgeniya Sysueva, Dmitry Nechaev, Vladimir Popov, Mikhail Tarakanov, and Alexander Supin (Inst. of Ecology and Evolution, 33 Leninskij prosp., Moscow 119071, Russia, popov.vl.vl@gmail.com)

Hearing thresholds were measured and audiograms were obtained in seven belugas, three males and four females, provisionally 2 to 7 years old. The measurements were performed using a transducer located 1 m in front of the head. The stimuli were tone pip trains of carrier frequencies ranging from 11.2 to 128 kHz with a pip rate of 1 kHz. Auditory evoked potentials (the rate following responses) were recorded from the head vertex. In majority of the subjects, audiograms were similar to the typical odontocete audiograms with the lowest thresholds (from 53.7 to 56.4 dB re 1 μPa) at mid-frequency range (from 32 to 64 kHz) and a sharp thresholds rise (up to 79.5 dB re 1 μPa) at high frequencies (90–128 kHz). One beluga (female, 6–7 years old) featured an asymmetric hearing loss within a frequency range from 22.5 to 54 kHz. The reason for the loss is the subject for not defined. The evoked potential audiograms should be included into base screening of odontocete subjects involved in any kind of hearing research. Work supported by the Russian Science Foundation (project # 17-74-20107), the Russian Foundation for Basic Research and The Russian Geographic Society.]

MONDAY AFTERNOON, 7 MAY 2018 — GREENWAY A, 3:00 P.M. TO 4:40 P.M.

Session 1pAO

Acoustical Oceanography and Underwater Acoustics: Acoustics in Naturally Constrained Environments: Estuaries, Bays, Inlets, Fjords, and Rivers II

Andone C. Lavery, Cochair
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D. Benjamin Reeder, Cochair
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Invited Papers

3:00

1pAO1. Progress on the use of autonomous underwater vehicles to study estuaries. Craig L. McNeil, Andrey Shcherbina (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, cmcneil@apl.washington.edu), Marcia J. Isakson, Autumn N. Kidwell (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), D. Benjamin Reeder (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA), Andone C. Lavery, and W Rockwell Geyer (Appl. Ocean Phys. and Eng., Woods Whole Oceanographic Inst., Woods Hole, MA)

Estuaries are a challenging environment to use acoustically navigated Autonomous Underwater Vehicles (AUVs) due to highly variable currents, relatively shallow and variable bathymetry, large buoyancy changes, suspended sediments, marine biota, and bubble plumes. The benefit to using AUVs is their ability to perform repeat automated surveys and targeted sampling of features using either remote control or on-board redirects. Our REMUS 100 AUVs are equipped with up/down looking ADCPs, CTDs, and optical backscatter sensors. Our AUVs use Long Base Line (LBL) underwater navigation, with up to four transponders, and were recently equipped to carry broadband hydrophones. We will share our experience operating these AUVs in several estuaries to characterize variability in optical and acoustical backscatter associated with estuarine features of interest (fronts, river plume, and the salt-wedge). We will discuss how these features negatively impact AUV sampling via, for example, degraded underwater communications and bottom tracking. We will also present some examples of concurrent sampling by AUVs and an advanced sonar (static and mobile) demonstrating the potential use of AUVs in estuarine research for 4D visualization of estuarine features of interest. [This work was supported by Office of Naval Research.]

3:20

1pAO2. Effects of tidal forcing in an estuarine environment on high-frequency acoustics. Marcia J. Isakson, Autumn N. Kidwell, and Mark Story (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

Tidal forcing in rivers and estuaries has a large impact on the use high-frequency acoustics for bathymetric mapping. Changes in the sound speed profile due to changing salinity and temperature conditions affect acoustic propagation, and bubbles, entrained in the tidal flow, cause high backscattering and attenuation. In this study, a forward-looking, high-frequency bathymetric sonar was used to observe estuarine dynamics near the mouth of the Connecticut River in June 2017. The sonar was deployed on a REMUS 600 AUV under the fresh water plume near the river mouth during flood tide, in the lower estuary during ebb tide and in the upper estuary during slack tide. Backscattering from subducted bubbles was observed which was dependent on the propagation path through the fresh or salt water. The bubble backscatter over a 200-m swath at the front of the fresh water plume allowed the structure of the plume to be observed from beneath the surface. These data are compared with other measurements on site, including X-Band radar and *in situ* salinity and temperature measurements. The data can be used to understand both acoustic propagation and estuarine dynamics in constrained environments. [Work sponsored by the Office of Naval Research.]

Contributed Papers

3:40

1pAO3. 3D underwater sound propagation in coastal 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), D. Benjamin Reeder (Naval Postgrad. School, Monterey, CA), and David K. Ralston (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Underwater sound propagation in a coastal ocean can be influenced by a variety of physical oceanographic conditions that are unique in the coastal and estuarine environment, such as salt wedge fronts, river-plume induced nonlinear internal gravity waves, strong tidal currents, surf zone waves, etc. The confined boundaries in the area can also produce reflection, diffraction and even scattering of sound due to shoaling seafloor. The morphological dynamics in the area is complex and results in corrugated and channeled bathymetry, which may cause 3D ducting and trapping of sound. In this talk, we will present our numerical study of 3D underwater sound propagation in Long Island Sound, a tidal estuary of the Atlantic Ocean. The Unstructured Grid Finite Volume Community Ocean Model (FVCOM) will be utilized to simulate physical oceanographic processes and produce water-column environmental input to the sound propagation model. A bathymetric database from high-resolution multibeam surveys will also be used to incorporate the realistic seafloor depth in the model. The numerical model results will be analyzed to present the evidences of 3D sound propagation effects and investigate the causes.

3:55

1pAO4. Acoustic propagation in the Connecticut River at low-to-mid frequencies. D. Benjamin Reeder, John E. Joseph, and Tarry Rago (Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, SP-311B, Monterey, CA 93943, dbreeder@nps.edu)

An estuary is a constrained environment which often hosts a salt wedge, the stratification of which is a function of the tide's range and speed of advance, river discharge volumetric flow rate and river mouth morphology. A field experiment was carried out in the Connecticut River in June 2017, one goal of which was to investigate the low-to-mid-frequency acoustic propagation characteristics of the riverine salt wedge as well as the plume outside the river mouth. Linear frequency-modulated (LFM) acoustic signals in the 500–2000 Hz band were collected during several tidal cycles. Data-model comparisons demonstrate the degree to which this highly energetic environment impacts acoustic propagation; dominant mechanisms are sound speed stratification, boundary interaction, flow noise, and background noise.

4:10

1pAO5. High-frequency acoustic observations of a tidal cycle in the Connecticut River. Autumn N. Kidwell, Marcia J. Isakson, Mark Story, and Andrew J. Reiter (Appl. Res. Labs., Univ. of Texas, Appl. Res. Labs., University of Texas, 10000 Burnet Rd., Austin, TX 78758, akidwell@arlut.utexas.edu)

Rivers and estuaries can be highly energetic and present a challenging environment for sonar deployment and observation. To better understand this environment, a high-frequency, wide-sector forward-looking bathymetric sonar was used to observe tidal phenomena in the lower Connecticut River in June 2017. The stationary sonar system provided Eulerian measurements of the lower estuary for a complete tidal cycle, covering a large portion of the main channel. The system was able to map the observed region and differentiate between bathymetry and non-stationary phenomena in the water column. The data highlight the difference in acoustic propagation for the different phases of the tidal cycle. Doppler processing of the data shows the motion of various subsurface features in the river. These data are compared with other observations, including simultaneous *in situ* point measurements of salinity, temperature and turbidity from a REMUS 100 AUV. The data from this experiment can be used to improve the understanding of acoustic propagation in riverine and estuarine environments. [Work sponsored by the Office of Naval Research.]

4:25

1pAO6. Spectral quantification of a tidal plume from an autonomous underwater vehicle. Andone C. Lavery (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 98 Water St., MS 11, Bigelow 211, Woods Hole, MA 02536, alavery@whoi.edu), Joseph T. Jurisa (Dept. of Civil and Environ. Eng., Portland State Univ., Portland, OR), and W. Rockwell Geyer (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Buoyant plumes are ubiquitous in estuarine environments, yet their frontal structure and dynamics are not fully understood. Tidal plumes are dynamic over a broad range of temporal and spatial scales, and are characterized by strongly stratified turbulence and frontal instabilities. Observations from autonomous underwater vehicles (AUVs) with a suite of sensors, including temperature, salinity, and acoustic backscatter, provide a useful complement to ship-board sampling to characterize the frontal structure and dynamics at high resolution. Results are presented from an experiment in which a broadband acoustic backscattering system (160–410 kHz) was integrated onto a REMUS-100 AUV and used to sample the near-field ebb tide plume at the mouth of the Connecticut River. Scattering mechanisms are determined from the frequency-dependent acoustic spectra, and acoustic measurements of the frontal structure are compared to data collected with a towed CTD array. Data are also presented that illustrate the challenges associated with AUV operations in estuarine environments, including high currents, shallow and variable bathymetry, large buoyancy changes, and bubble entrainment at the plume front.

Session 1pBA

Biomedical Acoustics: Transcranial Focused Ultrasound for Targeted Brain Therapies

Emad S. Ebbini, Chair
Electrical and Computer Engineering, University of Minnesota, 200 Union St. SE, Minneapolis, MN 55455

Chair's Introduction—1:25

Invited Papers

1:30

1pBA1. Progress in the utilization of focused ultrasound for treatment of the brain. Kullervo Hynynen (Medical Biophys., Univ. of Toronto, Sunnybrook Health Sci. Ctr., Toronto, ON M4N 3M5, Canada, khynynen@sri.utoronto.ca)

Magnetic Resonance imaging (MRI) guided focused ultrasound (FUS) has been shown to be able to safely ablate deep tissues in the brain in clinical settings resulting in reduction in recovery time and complications. Pre-clinical work has demonstrated that FUS energy when combined with micro bubbles(MB) could be used also for image-guided drug delivery into tumours and brain. The preclinical studies have shown significant survival benefits when FUS + MBs has been used to open the blood-brain-barrier (BBB) for large molecular chemotherapy agents, antibodies, and natural killer cells. Effective and safe delivery of nanoparticles and viral vectors has also been shown. This talk will briefly review the physical and engineering aspects related to the clinical FUS + MB exposures. The first ongoing clinical trials investigating the feasibility and safety of FUS + MB enhanced BBB permeability will be reviewed. Finally, the next generation phased array technology and the ultimate clinical potential of the FUS + MBs in brain treatments will be discussed.

1:55

1pBA2. Real-time monitoring of transcranial focused ultrasound *in vivo*. Dalong Liu and Emad S. Ebbini (Elec. and Comput. Eng., Univ. of Minnesota, 200 Union St. SE, Minneapolis, MN 55455, ebbin001@umn.edu)

Transcranial focused ultrasound (tFUS) is being investigated in a variety of medical applications. Both thermal and mechanical bioeffects of tFUS are of interest, but localization of the tFUS therapeutic beam remains a major concern in the presence of beam distortion through the skull. We have developed a system for monitoring and delivery of tFUS using dual-mode ultrasound array (DMUA) transducers. The system is capable of delivering therapeutic tFUS in pulsed mode with imaging pulses interleaved at frame rates up to 500 fps utilizing the same array elements. In this way, the DMUA provides inherent registration between the imaging and therapy system thus providing feedback from the target region as well as tissue structures in the path of the beam. The beamformed DMUA echo data was used to image the thermal and mechanical tissue response to the tFUS beam with high spatial specificity. *In vivo* data from small-animal experiments have shown a temperature sensitivity of 0.2°C with high spatial specificity. Speckle tracking at up to 500 fps also allowed for tracking mechanical deformation produced by the tFUS beam in the presence of pulsations and other brain tissue motions. These results demonstrate the feasibility of characterizing both mechanical and thermal effects of subtherapeutic tFUS beams *in vivo*. The results are significant in burgeoning applications where the neuromodulatory response is induced by a combination of thermal and mechanical interactions.

Contributed Papers

2:20

1pBA3. Transcranial acoustic cavitation localization with ultrafast power cavitation imaging in non-human primates. Mark T. Burgess, Maria E. Karakatsani, Iason Apostolakis, and Elisa Konofagou (Biomedical Eng., Columbia Univ., 630 West 168th St., Physicians and Surgeons 19-418, New York City, NY 10032, mtb2153@columbia.edu)

Acoustic cavitation-guided blood-brain barrier (BBB) opening with focused ultrasound (FUS) and microbubbles is a promising technique for safe and controlled opening of the BBB. Passive cavitation imaging has the ability to monitor the spatial intensity of acoustic cavitation for targeting verification and treatment monitoring. However, isolating acoustic cavitation emissions from tissue and skull reflections is a major challenge. In this study, we perform transcranial passive cavitation imaging with a 1.5D imaging array (M5Sc-D, bandwidth: 1.5–4 MHz, GE Medical Systems) placed in the central opening of a 0.5 MHz FUS transducer (H204, Sonic Concepts) in non-human primates. Broadband FUS pulses were used along with synchronous transmit and receive sequences to perform delay and sum beamforming with absolute time delays. Image sets were acquired at ultrafast frame rates (>1000 frames per second) for calculation of mean intensity images, i.e., power cavitation images. Spatiotemporal clutter filtering of image sets was explored over a range of FUS peak negative pressures (0.05–0.75 MPa) to increase power cavitation image sensitivity. Future work will explore the use of this method for safe and controlled BBB opening based on power cavitation image feedback. [Work supported in part by NIH grants R01AG038961 and R01EB009041.]

2:35

1pBA4. Wideband transskull refocusing of ultrasound beams using dual-mode ultrasound arrays: *Ex vivo* results. Hasan Aldiabat (Elec. and Comput. Eng., Univ. of Minnesota, 200 Union St SE, Minneapolis, MN 55455, aldia001@umn.edu), Parker D. O'Brien (Elec. and Comput. Eng., Univ. of Minnesota, Richfield, MN), Dalong Liu, and Emad S. Ebbini (Elec. and Comput. Eng., Univ. of Minnesota, Minneapolis, MN)

Transcranial focused ultrasound (tFUS) is capable of providing subtherapeutic and therapeutic ablative treatments for a variety of brain disorders. A major challenge towards widespread use of tFUS-based therapies stems from the complexity of the skull that could result in severe loss of focusing gain. Using extensive hydrophone scan measurements in plain water as well as transskull, we have documented a range of tFUS beam distortions for a variety of target points and access angles. In this paper, we present quantitative measurements of tFUS distortions due to skull aberrations at different operating frequencies. In addition, refocusing results for a variety of target points at different frequencies within the transducer bandwidth are presented in terms of improvement in focusing gain. Dual-mode ultrasound array (DMUA) prototype (64 elements, concave with 40-mm radius of curvature) was used. Skull samples were extracted from animal subjects that have undergone tFUS treatments using the DMUA prototype were utilized. Experiments were performed at a set of 31 discrete frequencies in the range 2.0 MHz–5.0 MHz. A needle hydrophone was used to measure the pressure waveforms at the target locations. The element transmission efficiency varied as a function of frequency in a nonmonotonic manner with a range of 5–15 dB variation for the different target points. The array focusing gain also varied nonmonotonically suggesting the need for broadband refocusing.

2:50–3:10 Panel Discussion

3:10–3:25 Break

Invited Papers

3:25

1pBA5. Transcranial focused ultrasound with multielement arrays: Decreasing the number of transducers without compromising the focusing quality. Jean-Francois Aubry (Institut Langevin, CNRS, 17 rue Moreau, Paris 75012, France, jean-francois.aubry@espci.fr)

Transcranial ultrasonic brain therapy at frequencies higher than 500 kHz requires adaptive focusing to compensate for the aberrations induced by the skull bone. This can be achieved by using multi-element arrays driven by a dedicated electronics. A growing number of elements was used to improve the focusing: 64 elements in 2000 [1], 300 in 2003 [2], 1024 in 2013 [3], and more to come. We will present some of the salient results obtained pre-clinically and clinically with such multi-element transcranial devices. Nevertheless, we will show that comparable transcranial focusing can be achieved with a novel approach in rupture with the current trend. It consists in a single-element transducer covered with a 3D silicone acoustic lens of variable and controlled thickness. Similar lenses have been introduced in the past to perform multiple focusing patterns in homogenous media [4] but recent 3d printing capabilities make 3D lenses a feasible option for transcranial adaptive focusing. We will show that the use of a lenses increases the acoustic intensity by 97 $\pm$ 56% on average, on three human skulls. [1] G. Clement *et al*., "A hemisphere array for non-invasive ultrasound brain therapy and surgery," Phys. Med. Biol. 2000. [2] M. Pernot *et al*., "High power transcranial beam steering for ultrasonic brain therapy," Phys. Med. Biol. 2003. [3] N. Lipsman *et al*., "MR-guided focused ultrasound thalamotomy for essential tremor: a proof-of-concept study," *The Lancet Neurology* 2013. [4] T. Fjield *et al*., "Low-profile lenses for ultrasound surgery," Phys. Med. Biol. 1999.

3:50

1pBA6. Focused ultrasound for modulation of the central and peripheral nervous system. Elisa Konofagou (Columbia Univ., 1210 Amsterdam Ave., ET351, New York, NY 10027, ek2191@columbia.edu)

Noninvasive neuromodulation has been the preferred option of neurological treatment but noninvasive approaches fall short when it comes to depth penetration. Ultrasound modulation has been shown feasible in several species including humans both *in vitro* and *in vivo*. In this paper, an overview of our group's ultrasound neuromodulation in both the central (CNS) and the peripheral (PNS) nervous systems will be provided. In CNS, both motor- and sensory-related responses have been elicited in mice *in vivo* both in ipsilateral and contralateral limbs and pupils, respectively. The success are was highly correlated with the applied intensity and pressure in both the limb movement and ocular changes. The brain regions targeted were the somatosensory and visual cortex for the limb movement and the superior colliculus and locus coeruleus for the pupil dilation. In PNS, stimulation and inhibition of the sciatic nerve with FUS was elicited at different ultrasound parameters *in vivo*. Displacement of the nerve highly correlated with the elicited motor response. The success rate also correlated with higher intensities. Histology confirmed safety while temperature elevation was more correlated with inhibitory responses.

4:15

1pBA7. Intranasal administration of temozolomide combined with focused ultrasound to enhance the survival of mice with glioma (A Pilot Study). Dezhuang Ye (Mech. Eng. and Material Sci., Washington Univ. in St. Louis, 1155 Claytonia Terrace #2N, 2N, Saint Louissaint Louis, MO 63117, dezhuang.ye@wustl.edu), Yimei Yue, Lifei Zhu, Hong Chen (Biomedical Eng., Wustl, St. Louis, MO),

Intranasal route provides therapeutics direct access to the brain through the nose-to-brain pathway, bypassing the BBB and minimizing systemic exposure. However, nasal delivery is limited by its low delivery efficiency and non-localized delivery. Focused ultrasound-enhanced intranasal (FUSIN) delivery is a new technique for noninvasive and localized delivery of therapeutic agents to the brain. Previous studies have shown its feasibility in the delivery of dextrans and brain-derived neurotrophic factor to the brain. The goal of this study was to demonstrate the application of FUSIN for the delivery of temozolomide (TMZ), a first-line drug for treating glioma tumors, for the treatment of glioblastoma in an orthotopic mouse model. U87 glioblastoma cells were trancranially implanted to nude mice. The tumor mice were divided into four groups: (1) FUSIN + TMZ (n = 7); (2) IN TMZ (n = 6); (3) Oral TMZ (n = 6); and (4) IN vehicle (n = 6), and were treated once each week for 4 weeks. Mice were weighed once a week. A body weight loss reaching 20% was selected as the endpoint criteria for the study. Mice in FUSIN group showed statistically significant higher survival rate when compared with mice in all other groups. This pilot study suggested that FUSIN can potentially be an effective technique for the treatment of brain diseases.

4:30

1pBA8. Broadband transskull multi-focus wavefront synthesis. Parker D. O'Brien (Elec. and Comput. Eng., Univ. of Minnesota–Twin Cities, 7710 Penn Ave., S, Apt. B318, Richfield, MN 55423, obrie844@umn.edu), Dalong Liu, and Emad S. Ebbini (Elec. and Comput. Eng., Univ. of Minnesota–Twin Cities, Minneapolis, MN)

Multiple-focus pattern synthesis using ultrasound arrays has been previously demonstrated. Furthermore, an optimal synthesis method was shown to provide well-behaved solutions. This method creates predefined focus patterns at a single frequency for an ultrasonic array operating in continuous wave. In this paper, we extend the approach to the broadband case using orthogonal frequency-division multiplexing (OFDM). A broadband multiple-focus pattern is formed by summing the results of the synthesis algorithm at a set of discrete frequencies within the transducer bandwidth. The OFDM approach allows for achieving a predictable focusing gain at the target point(s) due to the waveform orthogonality. Outside the target region, the different frequency components can be expected to provide a larger degree of destructive interference. Hydrophone scans of multiple-focus patterns in plain water as well as transskull were performed using a phased array with a multichannel arbitrary waveform generator. The array has a 3.5 MHz center frequency with a 6-dB bandwidth in the range 2.2 MHz–4.8 MHz. The skull samples were obtained from 300 to 350 g rats that were used in other *in vivo* experiments. The results show greater focus resolution than for any single frequency pattern while reducing the interference patterns outside the target region.

4:45

1pBA9. Acoustical experimental design for ultrasonic neurostimulation on rodents. Shane W. Lani (Johns Hopkins Appl. Phys. Lab, 1454 Catherine St., Decatur, GA 30030, lani.shane@gatech.edu), Marina Congedo, Grace Hwang, and Allan Rosenberg (Johns Hopkins Appl. Phys. Lab, Laurel, MD)

While the mechanisms of ultrasonic neurostimulation are still unknown, this method has been successfully shown on rodents from a variety of research groups. While the results are compelling, there has been limited work in characterizing the acoustic fields that are generated and ultimately delivered to the rodent brain which is important in order to determine the underlying neurostimulation mechanism. This presentation details some of the acoustical experimental design processes that are being undertaken at JHU/APL as well as our current progress in the rodent ultrasonic neurostimulation. The experimental design utilizes a complementary process of both simulation and experimentation verification to determine the expected delivered acoustic wave field. The simulation encompasses the full propagation path from the single element focused transducer, through an acoustic waveguide and skull, to the treatment location. This work uses a focused transducer with a center frequency of 0.5 MHz with several different waveguides that are characterized in our calibration tank with regards to intensity over space. Each of these waveguides is modeled and compared relative to the focal spot size and intensity delivered to the focal spot. These measurements are used to select an optimal waveguide which is used in the model to determine the expected performance of the full system. The presented results compare the performance of the waveguides and their impact on ultrasonic delivery to the rodent brain.

Session 1pID

Interdisciplinary: Introductions to Technical Committees

Jonathan R. Weber, Cochair
Durham School of Architectural Engineering & Construction, University of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816

Tao Sun, Cochair
Radiology, Brigham and Women's Hospital; Harvard Medical School; Tufts University, 221 Longwood Avenue, EBRC 514, Focused Ultrasound Laboratory, Boston, MA 02115

Vahid Naderyan, Cochair
Physics/National Center for Physical Acoustics, University of Mississippi, NCPA, 1 Coliseum Drive, University, MS 38677

Invited Papers

1:00

1pID1. The Technical Committee on Musical Acoustics: Diverse membership with a common interest. James P. Cottingham (Phys., Coe College, 1220 First Ave. NE, Cedar Rapids, IA 52402, jcotting@coe.edu)

The Technical Committee on Musical Acoustics is concerned with the application of science and technology to the field of music. Topics of current interest include the physics of musical sound production, music perception and cognition, and analysis and synthesis of musical sounds. TCMU is an interdisciplinary committee with ties to architectural, psychological and physiological, signal processing, speech, physical, and structural acoustics. Members of the committee represent a variety of backgrounds and interests, yet are united by a common interest in understanding all aspects of the science of music.

1:10

1pID2. An introduction to the Engineering Acoustics Technical Committee. Michael V. Scanlon (Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783-1197, michael.v.scanlon2.civ@mail.mil)

The Engineering Acoustics Technical Committee's scope encompasses the theory and practice of creating tools for investigating acoustical phenomena and applying knowledge of acoustics to practical utility. This includes the design and modeling of acoustical and vibrational transducers, arrays, and transduction systems in all media and frequency ranges; instrumentation, metrology, and calibration; measurement and computational techniques as they relate to acoustical phenomena and their utility; and the engineering of materials and devices. The breadth and scope of EATC's mission is an enabler to the other committees and to the interests of every member.

1:20

1pID3. Introduction to the Technical Committee on Physical Acoustics. Veerle M. Keppens (Dept. Mater. Sci. and Eng., Univ. of Tennessee, Knoxville, TN 37996, vkeppens@utk.edu)

The Technical Committee on Physical Acoustics (TCPA) of the Acoustical Society of America (ASA) is home to scientists and engineers with an interest in the underlying physics of acoustical phenomena and/or use acoustic waves to study the physical properties of matter. It is one of the broader communities within the Acoustical Society, as the research areas and methods cover the entire frequency range from infrasound to ultrasound, while studying the interaction of these sound waves in all three states of matter: liquids, solids, and gasses. TCPA currently has about 575 members with a primary interest in physical acoustics with 75–125 typically attending open TCPA meetings.

1:30

1pID4. An introduction to research topics in underwater acoustics. Jason D. Sagers (Environ. Sci. Lab., Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, sagers@arlut.utexas.edu)

The Acoustical Society of America Technical Committee on Underwater Acoustics (UW) investigates sound wave phenomena in marine environments, including oceans, lakes, and rivers. Research interests span a broad spectrum from the measurement and modeling of acoustic propagation and scattering, to the detection and characterization of underwater sound, to signal processing algorithms and statistics. This diverse technical committee also shares interests with Animal Bioacoustics (AB), Acoustical Oceanography (AO), and Signal Processing (SP). This talk will highlight a few past and present research topics in underwater acoustics, emphasizing the important role of sound as a tool in subsea research and exploration.

1:40

1pID5. An introduction to the acoustical oceanography technical committee. John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, jacolosi@nps.edu)

As is evidenced by the complex acoustic physiology and sophisticated auditory processing capability of marine organisms, it is clear that acoustics is a critical modality for interpreting the ocean environment. The acoustical oceanography (AO) technical committee (TC) seeks to foster a broad range of work in pure and applied acoustics with an aim towards gaining new fundamental understanding of physical, biological, geophysical, and chemical processes in the ocean and at its boundaries. With this focus, the AO TC is necessarily strongly interdisciplinary having close ties to other TCs such as animal bioacoustics, physical acoustics, signal processing, and underwater acoustics. This talk will describe many of the break-through discoveries that have been made in our field and point to inspiring present and future work.

1:50

1pID6. Overview of animal bioacoustics. Christine Erbe (Ctr. for Marine Sci. & Technol., Curtin Univ., Kent St., Bentley, WA 6102, Australia, c.erbe@curtin.edu.au)

Research in Animal Bioacoustics includes (1) animal sounds, communication, biosonar, and associated behavior; (2) sound production anatomy and neurophysiology; (3) auditory capacities and mechanisms, anatomy, and neurophysiology; (4) acoustic phylogeny, ontogeny, and cognition; (5) acoustic ecology, acoustic characterization of habitats, and effects of sound on animals; (6) passive acoustic tools and methods, hardware and software, for detection, classification, localization, tracking, density estimation, and behavior monitoring; and (7) active acoustic tools and methods, including animal-tracking sonars and echosounders, acoustic tags, pingers, deterrent devices, etc. Study species include birds, terrestrial mammals (in particular, bats), marine mammals, amphibians, reptiles, fishes, insects, and crustaceans. Researchers have diverse backgrounds ranging from acoustics, engineering, mathematics, and physics, to biology and zoology. We pool our expertise in this truly interdisciplinary field of Animal Bioacoustics.

2:00

1pID7. Biomedical acoustics: From diagnostic imaging to treating brain disorders. Subha Maruvada (U.S. Food and Drug Administration, 10903 New Hampshire Ave., Bldg. WO 62-2222, Silver Spring, MD 20993, subha.maruvada@fda.hhs.gov)

The Technical Committee on Biomedical Acoustics (BATC) is one of the most diverse groups in the Acoustical Society of America. BATC is comprised of scientists and engineers who study a wide range of biomedical applications using ultrasound from imaging fetuses to ablating fibroids and treating brain disorders. Diagnostic ultrasound has expanded to include exciting new techniques such as shear wave elastography and opto-acoustic imaging. Therapeutic ultrasound applications include physiotherapy, lithotripsy, as well as the treatment of tumors via thermal ablation, i.e., HITU, or mechanical ablation, i.e., histotripsy. The most recent developments in therapeutic ultrasound involve neuromodulation and transcranial magnetic resonance guided focused ultrasound for the treatment of various brain disorders, including essential tremor, neuropathic pain, and Parkinson's disease. The ability to temporarily open the blood-brain barrier using focused ultrasound has been demonstrated in animals and, more recently, in humans. Opening of the blood-brain barrier could allow the transmission of drugs to treat brain diseases. Diagnostic and therapeutic ultrasound applications continue to advance the field of medicine and are integral to the advancement of both diagnosis and treatment of debilitating diseases.

2:10

1pID8. Introduction to the structural acoustics and vibration technical committee. Robert M. Koch (Chief Technol. Office, Naval Undersea Warfare Ctr., Code 1176 Howell St., Bldg. 1346/4, Code 01CTO, Newport, RI 02841-1708, Robert.M.Koch@navy.mil)

The Structural Acoustics and Vibration (SAV) Technical Committee is a broad group of scientists and engineers sharing a common interest and shared stewardship in the health and advancement of the SAV technical discipline. Structural acoustics may be defined as a multidisciplinary coupled field of physics usually referring to the characterization of either (1) the sound power emitted by a vibrating structure subjected to external dynamic excitation or (2) the vibrational response of structures excited by incident sound fields or fluid excitation. While the fundamental scientific study of the underlying SAV physics is certainly important to SAV practitioners, there is also interest on the practical application of the prediction, control, and potential reduction of the vibroacoustic response of given structural acoustic systems. This paper first provides a fundamental definition of the underlying physics behind the SAV technical discipline. The many categories and subdivisions within the general SAV area are then addressed, along with illustrations of the broad array of scientific and engineering real-world applications in the field. Following that, examples of interesting career options as well as exciting new research areas in SAV are presented. Finally, the paper concludes with information regarding the makeup, functioning, and administrative philosophy of the SAVTC.

2:20

1pID9. Introduction to the Technical Committee on Noise. William J. Murphy (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov)

Noise presents both a health hazard and a societal hazard. Health hazards due to noise have been around since the development of metal forging and forming. As modern society has developed, community noise has affected where and how we interact with the soundscapes where we live. The United States Environmental Protection Agency, Office of Noise Abatement and Control was responsible for noise regulations that rate the production and reduction of noise until it was defunded. Federal agencies such as the National Institute for Occupational Safety and Health, the Occupational Safety and Health Administration, and the Mine Safety and Health Administration have conducted research to develop regulations for occupational noise exposures. Recent community noise issues have focused on windfarms and the low-frequency noise that nearby residents might experience. This presentation will cover a wide range of noise-related topics and highlight efforts in standards for noise assessment and noise control.

2:30

1pID10. Architectural acoustics: From concert halls to classrooms. Ronald Freiheit (Wenger Corp., 555 Park Dr., Owatonna, MN 55060, ron.freiheit@wengercorp.com) and Matthew T. Neal (Graduate Program in Acoust., Penn State Univ., University Park, PA)

Architectural Acoustics (AA) finds its application in and around the built environment and, in doing so, touches and interacts with many different disciplines. Within ASA, the TCAA consistently hosts joint special sessions with the noise, musical acoustics, psychological and physiological acoustics, and signal processing technical committees. Apart from its involvement with special sessions, many unique subcommittees bring added diversity within architectural acoustics. These include subcommittees on classroom acoustics, speech privacy, green building acoustics, healthcare acoustics, building performance standards, and the concert hall research group (CHRG). The makeup of members in the AA group is quite varied. It includes those involved in education and research to those in consulting as well as those in industries that supply products to this segment. Along with published research which is typical of most technical committees, a large segment of the TCAA's accomplishments are based on the projects produced by consultants or the products provided by industries. The AA group has consistently been one of the larger groups of active participants. There is a unique balance between research, application, and industry support for this segment of acoustics, making the TCAA a great place to see new perspectives and make new connections with colleagues, friends, and mentors.

2:40

1pID11. Psychological and physiological acoustics: Responses to sounds in biological systems. 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)

The Technical Committee on Psychological and Physiological (P&P) acoustics is organized around the investigation and dissemination of information about responses to sound in humans and other species. Areas of interest include, but are not limited to the following topics: (1) perception and perceptual organization of simple and complex sounds, including speech; (2) anatomy and function of the auditory pathways, including all physical and biological responses to auditory stimulation; (3) hearing disorders, hearing loss, and auditory prostheses; (4) vibrotactile and vestibular sensation, and the interaction of hearing with other sensory modalities; (5) developmental, aging, learning, and plasticity effects in auditory function; and (5) theories and models of auditory processes. Examples will be given of hot topics in these areas as well as areas of overlap with other TCs in the ASA.

2:50

1pID12. It is like ESP without the E: An Introduction to the Speech Communication Technical Committee. Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu)

Speech communication (SC) is a quintessentially human behavior. Scholars in the SC Technical Committee conduct research on three broad areas: the production of speech, the perception of speech, and the machine processing of speech. We study people across the globe and across the entire lifespan. Our work includes people with and without speech, language, and hearing impairments. This brief talk will highlight three recent projects by SC TC members that illustrate the breadth of our work. We will review research on how speech style affects memory for speech, how ultrasound can be used to measure tongue movement during speech, and how speech synthesis can be used to estimate articulatory movements from acoustic records of speech. We hope that this presentation will encourage students and new members from our society's many technical committees to learn more about how our work in the SC TC complements research and scholarship in many different fields of acoustics.

3:00

1pID13. Signal processing everywhere you look. Zoi-Heleni Michalopoulou (Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd., Newark, NJ 07102, michalop@njit.edu)

Whether designing an auditorium, interpreting a marine mammal song, analyzing a sonar signal, or extracting information from ambient noise, signal processing is there. With active research in the field, signal processing provides a multitude of tools that are critical for solutions of complex acoustic problems. With time-series analysis, time-frequency representations, higher-order statistics, model-based and Bayesian processing, compressed sensing, and many other approaches, acoustic signal processing brings together physics, mathematics, and statistics, theory and computation, for the better understanding of acoustical phenomena. The diversity of the field is evidenced by the fact that members of the Technical Committee on Signal Processing in Acoustics also have membership in a multitude of other technical committees: underwater acoustics, acoustical oceanography, animal bioacoustics, and architectural acoustics, to name a few. Additionally, many special sessions at Acoustical Society Meetings are cosponsored by the Technical Committee on Signal Processing in Acoustics, proof that signal processing is everywhere you look. It is this diverse field that we explore in this talk, providing an overview of current research in a number of areas.

Session 1pNS

Noise, Psychological and Physiological Acoustics, Signal Processing in Acoustics, Speech Communication, and Musical Acoustics: Open Source Audio Processing Platforms

Odile Clavier, Cochair
Creare, Inc., 16 Great Hollow Rd., Hanover, NH 03755

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—1:00

Invited Papers

1:05

1pNS1. SignalMaster—An open hardware and software system for speech and signal processing. Patrick Davies and Rafael E. Delgado (Intelligent Hearing Systems Corp, 6860 S.W. 81st St., Miami, FL 33143, pdavies@ihsys.com)

A portable speech and signal processing system was developed using a TMS320C6748 Digital Signal Processor (DSP) with two input and output channels with 32 bit Analog-to-Digital (AD) and Digital-to-Analog (DA) converters. Lithium-ion batteries provide up to 10 hours of freestanding operation. The system may also be used while connected to a personal computer (PC). Special boot loading code and an interface protocol was developed in order to allow communications and standardize the transfer of instructions and data between the DSP hardware and PC through a Universal Serial Bus (USB) interface cable. In addition, a PC-based user application interface high-level language dynamic link library was also developed to enable development of a wide range of applications. The library provides functions for user developed applications to communicate with the hardware and be able to upload C language programs into the DSP system. This allows for different programs to be uploaded in order to reconfigure the processing algorithms being executed for any experiment at any time. DSP and PC-based software source code examples provide users with the ability to develop their own user specific applications. Examples were developed to demonstrate the implementation of hearing aid amplification, filtering, and noise cancellation in real time.

1:25

1pNS2. Audiologic evaluation of the tympan open source hearing aid. Joshua M. Alexander (Speech, Lang., and Hearing Sci., Purdue Univ., Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907, alexan14@purdue.edu), Odile Clavier, and William Audette (Creare LLC, Hanover, NH)

The "Tympan" open source hearing aid (Creare LCC) was developed in response to needs identified by the 2014 Open Speech Signal Processing Platform Workshop hosted by NIH. The Tympan Phase I prototype is a neck-worn device with external microphones and headphones implemented on a Teensy 3.6 development board and Arduino software platform. Custom-designed electronics for audio input acquisition, digital signal processing, power management, and wireless communication allows expert users to create new algorithms directly in firmware. Operating in parallel, a custom software interface allows "occasional" users with the necessary support to modify easily parameters of existing algorithms. A variety of electroacoustic analyses, including throughput delay, amplitude compression input/output curves and time constants, harmonic distortion, internal noise, frequency response, and other real-ear measurements were performed on the Phase I prototype using commercial hearing aid analyzers and compared to today's premium hearing aid products. Overall, the results indicate that the Tympan Phase I prototype performs similarly to an actual hearing aid on these metrics. Results from an early Phase II prototype will also be reported. [Work supported by NIH NIDCD Grant #1R44DC015445-01 to Creare LLC.]

1:45

1pNS3. Tools for assessing efficacy of hearing loss compensation. Krishna Chaithanya Vastare (Dept. of Elec. and Comput. Eng., Univ. of California, San Diego, Atkinson Hall, 9500 Gilman Dr., La Jolla, CA 92093, kvastare@ucsd.edu), Sergio Luna (Dept. of Mathematics, Univ. of California, San Diego, La Jolla, CA), Tamara Zubaity (Dept. of Cognit. Sci., Univ. of California, San Diego, La Jolla, CA), Ganz Chockalingam (Qualcomm Inst., Univ. of California, San Diego, La Jolla, CA), and Harinath Garudadri (Dept. of Elec. and Comput. Eng., Univ. of California, San Diego, La Jolla, CA)

We are developing a real-time, wearable, Open-source Speech-Processing platform (OSP) that can be configured by audiologists and Hearing Aid (HA) researchers for lab and field studies. This contribution describes OSP tools to (i) control the HA state, such as amplification parameters in each subband, (ii) provide stimuli to the user, and (iii) get feedback from the user. The system is based on a web

server with interfaces to the HA state, the environment state (e.g., background noise characteristics, reverberation conditions, GPS location, etc.) and the user state (as inferred from ecological momentary assessments). We describe Application Programmer's Interfaces (APIs) used in Hypertext Markup Language (HTML) and server side scripting language (e.g., PHP) to create web applications aimed at assessing efficacy of various Hearing Loss (HL) compensation approaches. HTML and PHP are versatile tools and lot easier to develop functional skills compared with software tools used in the OSP. We provide example scripts to run tests such as A/B comparison, American version of the four alternative auditory feature test (AFAAF), etc. Audiologists and hearing scientists can modify these example scripts to create studies aimed at understanding the interactions between HA state, environment and user state and discover novel HL compensation approaches.

2:05

1pNS4. An open computational platform for low-latency real-time audio signal processing using field programmable gate arrays.
Ross K. Snider, Christopher N. Casebeer (ECE Dept., Montana State Univ., 610 Cobleigh Hall, Bozeman, MT 59717, ross.snider@montana.edu), and Raymond J. Weber (Flat Earth Inc., Bozeman, MT)

Field Programmable Gate Arrays (FPGAs) provide flexible computational architectures that are ideal for digital signal processing (DSP). With support of a NIH/NIDCD SBIR grant, we are developing an open FPGA platform for the speech, hearing, and acoustics research communities. The advantage of using FPGAs in a computational platform over conventional CPU approaches is the ability to implement low-latency high-performance signal processing with deterministic latencies. The hardware portion of the platform includes an audio codec and an Intel System-on-Chip (SoC) FPGA that contains ARM CPUs alongside the computational fabric that allows custom data plane designs. Development uses Mathwork's Simulink that allows exploration and simulation of signal processing algorithms. Once a Simulink model has been developed to implement audio processing in either the time domain or frequency domain, VHDL code can be generated that implements the desired signal processing. The VHDL code is then implemented in the FPGA computational fabric where the FPGA functions as a real-time signal processor. We are currently soliciting ideas/feedback from the speech, hearing, and acoustic communities as to what features they would like to see in the next iteration of the open FPGA-based computational platform. [NIH/NIDCD R44DC015443: www.openspeechtools.com]

2:25

1pNS5. An open source noise dosimeter for evaluating exposure metrics. Christopher J. Smalt, Lawrence Thul, Shakti K. Davis, and Paul Calamia (MIT Lincoln Lab., 244 Wood St, Lexington, MA 02420, Christopher.Smalt@ll.mit.edu)

Noise dosimeters can be effective tools for measuring individual risk to sound exposure, especially in free-moving environments where sound pressure levels can vary significantly as a function of location and time. Commercial noise dosimeters typically compute noise exposure as an average A-weighted energy over an 8-hour work day (LAeq8hr), and health guidelines suggest a maximum exposure of 85 dB or 90 dB to limit noise-induced hearing injuries. However, recent studies have suggested that conventional noise exposure metrics may not be satisfactory and may not predict susceptibility to synaptic damage in the cochlea nor protect suprathreshold hearing. There is also evidence to suggest that the LAeq may under-predict health risk in complex noise environments, where there is both continuous and impulse noise, and over-predict health risk for long-duration impulse noise. This suggests a need for improved dosimeters with the ability to incorporate new noise metrics. Here we utilize a low-cost open source audio platform to implement standard and experimental noise exposure metrics in an open-source microcontroller programming language. A Bluetooth connection also enables real-time reporting of noise exposures to a smartphone. This system has a significant advantage over using persistent acoustic recordings which have privacy issues and may capture conversations. We envision that this system will be used to further noise exposure research and help evaluate new metrics of auditory health risk. [Work supported by ONR.]

Contributed Paper

2:45

1pNS6. Recent developments in the NASA auralization framework.
Brian C. Tuttle (Analytical Mech. Assoc., Inc., M.S. 463, NASA Langley Res. Ctr., Hampton, VA 23681-2199, btuttle1@gmail.com), Aric R. Aumann (Sci. Applications Int. Corp., Inc., Hampton, VA), and Stephen A. Rizzi (AeroAcoust. Branch, NASA Langley Res. Ctr., Hampton, VA)

The NASA Auralization Framework (NAF) is an open software architecture that facilitates the conversion of numerical noise data, specifically aircraft flyover noise, into an audible pressure-time history that may be further analyzed or used as stimuli in psychoacoustic studies. The framework provides a set of libraries for management of the auditory scene and propagation paths, synthesis of simple sources, and a basic environment including a uniform standard atmosphere and hard flat ground. Advanced capabilities may be added in the form of dynamic-link library plugins either developed by the user or provided by NASA as pre-compiled binaries. Recent additions to the NASA collection of advanced plugins include synthesis of periodic noise sources, enhanced ground reflection and impedance models, an atmospheric turbulence model, a post-processing module for computing sound quality metrics, and an interface with the Aircraft NOise Prediction Program 2 (ANOPP2). This presentation highlights these advanced capabilities as they relate to current NASA research into noise from existing and proposed aircraft designs.

3:00–3:15 Break

Invited Papers

3:15

1pNS7. Open portable platform for hearing aid research. Caslav Pavlovic (BatAndCat Corp., 602 Hawthorne Ave., Palo Alto, CA 94301, chas@batandcat.com), Volker Hohmann, Hendrik Kayser (Univ. of Oldenburg and HörTech gGmbh, Oldenburg, Germany), Louis Wong (BatAndCat Corp., Cambell, CA), Tobias Herzke (Univ. of Oldenburg and HörTech gGmbh, Oldenburg, Germany), S. R. Prakash, zezhang Hou (BatAndCat Corp., Palo Alto, CA), and Paul Maanen (Univ. of Oldenburg and HörTech gGmbh, Oldenburgh, Germany)

The NIDCD has recently funded a number of projects to develop portable signal processing tools that enable real-time processing of the acoustic environment. The overarching goal is to provide a large group of researchers with the means to efficiently develop and evaluate, in collaborative multi-center environments, novel signal processing schemes, individualized fitting procedures, and technical solutions and services for hearing apparatus such as hearing aids and assistive listening devices. We report on the specific goals and results of two such projects. In one of them (R01DC015429), an open source software platform for real-time runtime environments is developed: The open Master Hearing Aid (openMHA). It provides an extendible set of algorithms for hearing aid signal processing and runs under Linux, Windows, and Mac operating systems on standard PC platforms and on small-scale ARM-based boards. An optimized version of openMHA is provided for the companion SBIR project (R44DC016247), which is a portable, rigid, versatile, and wearable platform featuring an ARM Cortex®-A8 processor. The resulting Portable Hearing Aid Community Platform consists of both hardware elements to provide the advanced desired functionality and software routines to provide for all the features that researchers may need to develop new algorithms.

3:35

1pNS8. Electroacoustic and behavioral evaluation of an open source audio processing platform. Daniel M. Rasetshwane, Judy G. Kopun, Ryan W. McCreery, Stephen T. Neely (Boys Town National Res. Hospital, 555 North 30th St, Omaha, NE 68131, daniel.rasetshwane@boystown.org), Marc A. Brennan (Univ. of Nebraska-Lincoln, Omaha, NE), William Audette, and Odile Clavier (Creare LLC, Hanover, NH)

Hearing-aid (HA) research in academia is limited by the lack of wearable, reconfigurable, and reprogrammable audio processing platforms. We are developing such a platform, called the Tympan, which includes a Teensy 3.6 processor board that leverages the Arduino development environment while providing powerful computational capabilities. Custom-designed electronics are used for audio control, power management, and wireless communication. User-friendly software can be used to test the relative benefits of amplification variants. Users can also implement new algorithms and modify parameters of algorithms. This study evaluated an eight-channel HA implemented on the Tympan, relative to a commercially available HA. Gain was prescribed using NAL-NL1. Eighteen participants with hearing loss were tested. Electroacoustic tests included ANSI 3.22-2009 standard clinical measurements, the HA speech perception index, and the HA speech quality index. Behavioral tests included recognition of CASPA words in quiet and AzBio sentences in noise. The Tympan HA performed similar to the commercially-available HA on all tasks. Efforts are ongoing to miniaturize the Tympan hardware, increase flexibility of the software, and add features such as feedback management. The collaborative development and open sharing of algorithms facilitated by the Tympan will lead to advances in HA research and in other audio signal processing fields.

3:55

1pNS9. Open-source speech-processing platforms: An application example. Arthur Boothroyd (Speech, Lang., Hearing Sci., San Diego State Univ., Campanile Dr., San Diego, CA 92182), Harinath Garudadri (Qualcomm Inst., Univ. of California, San Diego, La Jolla, CA), and Gregory Hobbs (Speech, Lang., Hearing Sci., San Diego State Univ., Campanile Dr., San Diego, CA 92182, ghobbs@sdsu.edu)

Speedy assessment of frequency-importance functions for a variety of speech materials is possible with an Open-source Speech-Processing (OSP) platform. One such platform is that developed at the University of California San Diego. It provides real-time, six-band, processing of microphone input from ear-level transducer assemblies. This platform was used to provide normative data on a new consonant-contrast test developed for use in studies of hearing-aid self-fitting. Performance of young normally hearing adults was measured under various conditions of low-pass filtering. Because the UCSD system includes two sets of filters whose center and crossover frequencies differ by half an octave, it was possible to test with cut-off frequencies at half-octave intervals from 250 through 8000 Hz. Group mean composite contrast score, after correction for chance, fell to around 50% at a (6 dB) cut-off frequency of 660 Hz. With a cut-off frequency of 1 kHz, group-mean contrast scores were in the region of 30 to 35% for place and clustering, 70% for continuance, and over 90% for voicing. Determining the frequency-importance functions of various speech materials and speech features is just one example of the potential value of OSPs in hearing-science and hearing-aid research.

4:15

1pNS10. Smartphone as a research platform for hearing study and hearing aid applications. Issa M. S. Panahi, Nasser Kehtarnavaz (Elec. Eng., Univ. of Texas at Dallas, EC33, 800 West Campbell Rd., Richardson, TX 75080, issa.panahi@utdallas.edu), and Linda Thibodeau (School of Behavioral and Brain Sci., Univ. of Texas at Dallas, Richardson, TX)

Smartphones are widely available and used by many people. Portability, processing power and useful features of smartphone, such as audio input/output and wire/wireless connectivity, offer unique opportunity to use smartphone as an open source research platform capable of implementing in real time many computationally intensive adaptive signal processing algorithms aimed at improving hearing study and hearing aid applications. In this paper, we present an overview of several adaptive algorithms developed for classifying the background noise, suppressing the noise, and enhancing the speech in different noisy environments. The proposed algorithms for hearing aid applications can run on existing smartphones such as the Android-based and iOS-based smartphones in real time. Objective and subjective test results are presented illustrating the performance of proposed algorithms running on smartphones in real time.

4:35–5:00 Panel Discussion

Session 1pPA

Physical Acoustics, Structural Acoustics and Vibration, Architectural Acoustics, and Underwater Acoustics: Novel Methods in Computational Acoustics II

D. Keith Wilson, Cochair
Cold Regions Research and Engineering Laboratory, U.S. Army Engineer Research and Development Center, 72 Lyme Rd., Hanover, NH 03768-1290

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

Invited Papers

1:00

1pPA1. Target response in a focused, modulated sound field: Modeling and experiment. Timothy D. Daniel, Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., WSU, Pullman, WA 99164, timothy.daniel@email.wsu.edu), Ahmad T. Abawi (HLS, Res., La Jolla, CA), and Ivars P. Kirsteins (NUWC, Newport, RI)

In a previous work, we had shown that low frequency flexural modes of circular plates and cylinders could be excited using modulated radiation pressure generated by focused ultrasound [J. Acoust. Soc. Am. **139**, 2053 (2016)]. We recently conducted experiments probing how the response of these targets varied as a function of position in the focused ultrasound field. Surprisingly, the response of the circular plate was found to change sign as it was moved away from the source. Two different models were developed to analyze the target response and understand this sign change. A purely geometric model based on ray optics and a semi-physical optics model that uses the incident intensity calculated for the focused source, using a Rayleigh-Sommerfeld integral. Both geometric and semi-physical optics models predict a sign change for the plate response at approximately the correct distance from the source. The sign change is due to a factor in a mode projection that more strongly weights points farther from the center of the plate. The physical optics model was also applied to cylindrical targets. [Work supported by ONR.]

1:20

1pPA2. Potential of theoretical approach and T-matrix method for designing numerical acoustical tweezers toolbox. Zhixiong Gong (Dept. of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., Pullman, Washington), Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA), and Wei Li (Dept. of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., 1037 Luoyu Rd. Hongshan Dist., Wuhan, Hubei 430074, China, hustliw@hust.edu.cn)

Partial-wave series solution and T-matrix method (TMM) could be used for computations of radiation force and torque exerted on objects from an acoustic Bessel beam (ABB) with arbitrary order and location. The radiation force and torque in Bessel beams could be expressed in terms of incident and scattered coefficients which could be obtained analytically based on the multipole expansion method [Gong *et al*., J. Acoust. Soc. Am. **141**, EL574–578 (2017)].The investigation of radiation force and torque could be used to design numerical acoustical tweezers toolbox that may manipulate particles and cells as expected, showing potential in physical chemistry, biomedicine, and so on. This work claims the possibility and potential of the theoretical and numerical methods for acoustical tweezers toolbox which could trap, pull, push, and rotate targets of different shapes and components. Both the progressive and (quasi)standing Bessel beams will be considered. In addition, beams which could be expanded in terms of basis functions, for example, Gaussian beams, could also be included in this numerical toolbox. The numerical toolbox of acoustical tweezers could help to guide experimental set-ups. [Work supported by NSFC (W.L. and Z.G.), ONR (P.L.M.), and China Scholarship Council (Z.G.).]

1:40

1pPA3. A modal collocation algorithm for high-frequency propagation in discontinuous waveguides. Jerry H. Ginsberg (School of Mech. Eng., Georgia Inst. of Technol., 5661 Woodsong Dr., Dunwoody, GA 30338-2854, j.h.ginsberg@comcast.net)

A procedure for analyzing high-frequency propagation in adjoining constant width waveguides was presented previously [Ginsberg, JASA **134**, 4217 (2013)]. The parallel walls were taken to be rigid, and no results were provided. This presentation uses that development as the foundation for the analysis of waveguides having an arbitrary number of sections, each of whose walls have arbitrary local impedance. The analytical basis is description of the pressure field in terms of transverse pressure modes, each of which is associated with upstream and downstream propagation at wavenumbers that may be complex, depending on the wall properties. Pressure and particle velocity conditions at junctions of two segments are enforced exactly by collocation along the common cross section. Initiating the procedure at a passive termination leads to recursion relations for modal reflection and transmission coefficients at each junction, which are independent of the nature of the excitation. Satisfaction of the conditions at the end that is excited leads to another algorithm for all modal amplitudes. An example is used to compare the convergence properties of the collocation method to an analysis based on the orthogonality of the transverse modes.

2:00

1pPA4. A Green's function parabolic equation description of infrasound propagation in an inhomogeneous and moving atmosphere. Michael B. Muhlestein, Vladimir E. Ostashev, and D. K. Wilson (US Army ERDC-CRREL, 72 Lyme Rd., Hanover, NH 03755, Michael.B.Muhlestein@usace.army.mil)

Accurate atmospheric infrasound propagation models must account for variations in local sound speed and ambient flow. This paper describes and demonstrates a method to extend the Green's function parabolic equation (GFPE) to account for 3D high Mach number ambient flow in addition to an inhomogeneous atmosphere. Predictions of infrasonic propagation using the resulting GFPE model are then compared with predictions using other models.

2:20

1pPA5. Non-Markov character of sound propagation over relatively short ranges in the turbulent atmosphere. Vladimir E. Ostashev and D. Keith Wilson (U.S. Army Engineer Res. and Developmnet Ctr., 72 Lyme Rd., Hanover, NH 03755, vladimir.ostashev@colorado.edu)

The Markov approximation is widely used in wave propagation in random media, including sound propagation in the turbulent atmosphere. This approximation, which significantly simplifies formulations for the statistical moments of the propagating field, is valid if the propagation range is greater than the outer length scale of random inhomogeneities. For sound propagation in the turbulent atmosphere, this length scale can be as large as 400–500 m (23% of the height of the atmospheric boundary layer), indicating that the Markov approximation might not be applicable for relatively short ranges. In this paper, the statistical characteristics of a sound field propagating in the turbulent atmosphere, such as the correlation functions of the log-amplitude and phase fluctuations, the mean sound field, and the mutual coherence function are calculated, without using the Markov approximation. The new results are then compared with those obtained in the Markov approximation. The difference between the two formulations and the range of applicability of the Markov approximation are analyzed for different meteorological regimes of the atmospheric boundary layer characterized by the surface heat flux and the friction velocity.

2:40–2:55 Break

2:55

1pPA6. Exact wide-angle formulation of the Beilis-Tappert method. Kenneth E. Gilbert, Xiao Di (Physics/NCPA, Univ. of MS, P.O. box 35, 1703 Hunter Rd., University, MS 38871, kgilbert@olemiss.edu), and Carl R. Hart (U.S. Army Engineer Res. and Development Ctr., Cold Regions Res. and Eng. Lab., Hanover, NH)

The Beilis-Tappert method was originally developed for narrow-angle acoustic propagation under a rough sea surface [A. Beilis and F. D. Tappert, J. Acoust. Soc. Am. **66**, 811–826 (1979)]. The method has also been applied to narrow-angle propagation over irregular terrain for acoustic waves and radar. It is shown here that an exact wide-angle formulation of the Beilis-Tappert method can be derived simply by replacing $\partial/\partial z$ with $\partial/\partial z + ik_0 \tan\varphi$, where $k_0 = 2\pi/\lambda$, λ is the physical wavelength, φ is the slope angle, and $i= \sqrt{-1}$. The exact formulation makes clear that for large slope angles, much of the acoustic field does not propagate, but decays exponentially with range. Existing finite difference methods and all narrow-angle methods fail to properly account for the exponential decay with a range of the non-propagating components of the acoustic field. The exponential decay with this range is qualitatively explained in terms of rays and quantitatively explained in terms of waves. Properly accounting for the propagating and non-propagating components of the acoustic field is explained.

3:15

1pPA7. On the use of high-order time-domain impedance boundary conditions. Didier Dragna and Philippe Blanc-Benon (Ctr. Acoustique, Laboratoire de Mécanique des Fluides et d'Acoustique, Ecole Centrale de Lyon, 36 Ave. Guy de Collongue, Ecully 69134 Ecully Cedex, France, Philippe.blanc-benon@ec-lyon.fr)

Time-domain simulations are well-suited to study broadband sound propagation. One of the difficulties is the translation of frequency-dependent impedance boundary conditions that leads in the time domain to convolutions. Several methods have been proposed in the literature to have an efficient computation of convolutions. They are based on a multipole approximation of the impedance and on approximations of time-variations of the acoustic variables, allowing the convolution to be simply evaluated by recursive relations. Recently, Dragna *et al.* [J. Acoust. Soc. Am. **138**, 1030–1042 (2015)] have highlighted that recursive convolution methods are low-order methods and have introduced a new method, referred to as the auxiliary differential equation method, which allows one to compute convolutions by integrating in time ordinary differential equations. Its main advantage is that it preserved the order of accuracy. This approach was employed in Troian *et al.* [J. Sound Vib. **392**, 200–216 (2017)] to derive a time-domain impedance boundary condition (TDBIC). This paper aims at evaluating the accuracy and efficiency of this novel TDBIC and to compare it to those of recursive convolution approaches. The novel TDBIC is first presented. A one-dimensional test case, dealing with reflection of an acoustic pulse over an absorbing wall is then investigated. Examples of simulations in three-dimensional geometries in the context of duct acoustics or outdoor sound propagation are then shown.

3:35

1pPA8. Computation of dispersion curves of guided waves using discontinuous Galerkin finite element method: Application to functionally graded materials. Salah-eddine Hebaz, Farouk Benmeddour, Emmanuel Moulin, Jamal Assaad, Marc Duquennoy, and Mohammadi Ouaftouh (Univ. Valenciennes, CNRS, Univ. Lille, YNCREA, Centrale Lille, UMR 8520 - IEMN, DOAE, 12 Ave. de l'industrie, Appart. A213, Valenciennes 59300, France, Salah-Eddine.Hebaz@univ-valenciennes.fr)

The study of dispersive behavior of guided waves is of great interest in non-destructive testing and development of inspection systems. In this work, a semi-analytical numerical formulation for the computation of dispersion curves and mode shapes in plate-like structures is presented. The formulation is based on a discontinuous Galerkin Finite Element Method. The discretisation of the ordinary differential system in the through-thickness direction yields an eigenvalue problem. The resolution of the latter for a frequency range provides the dispersion curves. The study focuses on Lamb modes in functionally graded material plates with traction-free boundary conditions. The influence of the gradient variations is demonstrated. First, a multilayered approximation consisting of homogeneous laminate is used. Afterwards, the gradients of the parameters are added to the formulation. Numerical examples are presented and compared with those found in the literature. The results are free of spurious modes and show an excellent agreement for all cases, specially for continuously varying properties. The method is very efficient and numerically stable for small and large wave numbers. It was found that a high accuracy is obtained when high-order elements are used on a relatively coarse mesh.

3:50

1pPA9. Investigation of broadband attenuation with fractal diffusers as noise barrier top edge devices using indirect boundary element method. Prachee Priyadarshinee (Mech. Eng., National Univ. of Singapore, #21-256B, 38 College Ave. East, Singapore 138601, Singapore, a0135595@u.nus.edu)

Computation of broadband noise barrier performance in three-dimension is computationally expensive. Indirect boundary element method with double layer potentials has been developed as a numerical tool in this acoustic study. It naturally permits noise barriers to be modelled as thin structures, thus reducing the computational cost when compared to the more abundant direct boundary element tools employed for noise barriers. The present numerical model has been validated with analytical solution. Its computational capabilities are compared with a direct boundary element tool, developed in-house. The applicability of this tool has been demonstrated with the study of fractal acoustic diffusers as noise barrier top-edge devices. Schroeder diffusers have been reported to provide good noise attenuation at their design frequency, when used as noise barrier top-edge devices. This study investigates the broadband performance of fractal diffusers as noise barrier top-edge devices.

4:05

1pPA10. Assessment of learning algorithms to model perception of sound. Menachem Rafaelof (NIA-NASA Langley Res. Ctr., M.S. 463, 100 Exploration Way, Hampton, VA 23681-2199, menachem.rafaelof@nasa.gov) and Andrew Schroeder (Acoust. Branch, NASA Langley Res. Ctr., Lancaster, OH)

Predicting human response to complex sound is a nontrivial task. Besides large differences among subjects and practically infinite types of stimuli, human response to sound is typically quantified by a few parameters having nonlinear behavior. Still, such predictions are valuable for the assessment of sound quality, which is a critical step toward the development of systems that offer improved human comfort, productivity and wellness. The overall objective of this work is to learn about human perception of sound through the use of machine learning algorithms. Learning algorithms are ideal for modeling the complex behavior of subjective parameters and identifying new trends with the potential to accumulate knowledge from different experiments. This work compares the performance of four learning algorithms (linear regression, support vector machines, decision trees, and random forests) to predict annoyance due to complex sound. Construction of these models relies on the annoyance response of 38 subjects to 103 sounds described by five known predictors (loudness, roughness, sharpness, tonality, and fluctuation strength). Comparison of these algorithms in terms of prediction accuracy, model interpretability, versatility, and computation time indicates that decision trees and random forests are the best algorithms for this task.

Session 1pPPa

Psychological and Physiological Acoustics and Signal Processing in Acoustics: Future Directions for Hearing Aids: Multi-Sensor, User-Informed, and Environment-Aware

Martin McKinney, Cochair
Starkey Hearing Technologies, 6600 Washington Ave. S, Eden Prairie, MN 55344

Tao Zhang, Cochair
Signal Processing Research, Starkey Hearing Technologies, 8602 Zachman Circle, Eden Prairie, MN 55344

Invited Papers

1:00

1pPPa1. Trends that are shaping the future of hearing aid technology. Brent Edwards (National Acoust. Labs., Australian Hearing Hub, Level 4, 16 University Ave., Macquarie Univ., NSW 2109, Australia, brent.edwards@nal.gov.au)

The development of hearing aid technology has accelerated over the past decade. Hearing aids are converging with consumer electronics in the area of hearables, and a recent government law mandating the creation of an over-the-counter hearing aid category will continue to bring a more consumer electronics focus to hearing aids. Meanwhile, new advances in hearing science are redefining the criteria of who needs hearing help. This talk will review these new intersections of technology and hearing need. It will also detail the technological and psychoacoustical challenges that face the ability of these new technologies to meet the needs of current hearing aid wearers, the needs of this emerging segment of the hearing impaired, and changes to hearing health delivery.

1:20

1pPPa2. Ecological momentary assessments for evaluation of hearing-aid preference. Karolina Smeds, Florian Wolters, Josefina Larsson, Petra Herrlin, and Martin Dahlquist (ORCA Europe, Widex A/S, Maria Bangata 4, Stockholm SE-118 63, Sweden, karolina.smeds@orca-eu.info)

Ecological Momentary Assessments (EMA) is a method that involves repeated evaluations in for instance a hearing-device user's everyday life. Compared to retrospective evaluations, the EMA method minimizes memory bias, and by making several evaluations each day, patterns in the data can be studied. In a pilot study at ORCA Europe, an EMA method was used to evaluate preference for two hearing-aid programs. For two weeks, the test participants were prompted to make evaluations, using a smartphone questionnaire. They were first asked to describe and classify each listening situation into categories such as "conversation with one person" and "monitoring surroundings." Then they made paired comparisons of the two hearing-aid programs. After the field-trial period, the participants also selected their overall preferred program. Group data for all listening situations pooled showed that the two programs were preferred almost equally often. However, when the data were divided into the seven listening categories, a pattern of preference could be seen. There also seemed to be an association between the test participants' auditory ecology profiles and their overall preferred program. The study will be presented, and the potential of the EMA method for research, development, clinical evaluations, and individualization of hearing-aid settings will be discussed.

1:40

1pPPa3. Listener preferences and speech recognition outcomes using self-adjusted hearing aid amplification. Trevor T. Perry (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, 164 Pillsbury Dr SE, Minneapolis, MN 55455, trevortperry@gmail.com), Peggy B. Nelson (Ctr. for Appl. and Translational Sensory Sci., Univ. of Minnesota, Minneapolis, MN), and Danne VanTasell (Ear Machine, Minneapolis, MN)

Self-adjustment of hearing aid gain empowers users to set amplification parameters according to their personal preferences. Listeners with mild-to-moderate hearing loss demonstrated self-consistency when adjusting gain in noisy backgrounds, although variability across subjects was very high and poorly predicted by user characteristics such as age, hearing thresholds, or previous experience using hearing aids. Despite the wide range of self-fit settings, speech intelligibility in noise was similar between self-fit and audiologist-fit settings. For average-level speech in noisy backgrounds, audibility was driven primarily by the signal-to-noise ratio, not gain settings. Self-adjustment for speech in quiet backgrounds will be compared to results obtained in noise. User interactions with the self-adjustment technology as well as preferences for self-fit settings and audiologist fit settings suggest that users can successfully use self-adjustment technology to achieve more desirable amplification from their hearing aids.

2:00

1pPPa4. Research on hearing-aid self-adjustment by adults. Carol Mackersie, Arthur Boothroyd (Speech, Lang. and Hearing Sci., San Diego State Univ., 2550 Brant St., SDSU: Speech, Lang, Hearing Sci., San Diego, CA 92101, cmackers@mail.sdsu.edu), and Harinath Garudadri (Qualcomm Inst., Univ. of California, San Diego, La Jolla, CA)

The purpose of the work is to develop a protocol for user self-adjustment of hearing aids and to determine its efficacy and candidacy. An initial study involved 26 adults with hearing loss. Control of overall volume, high-frequency boost, and low-frequency cut employed prerecorded and preprocessed sentence stimuli. Participants took a speech-perception test after an initial self-adjustment and then had the opportunity to repeat the adjustment. The final self-selected outputs were not significantly different from those prescribed by a widely used threshold-based method (NAL-NL2)—regardless of prior hearing-aid experience. All but one participant attained a speech-intelligibility index of 60%. Previous users of hearing aids, however, did not meet this criterion until after taking the speech-perception test. This work is continuing with the UCSD Open-source Speech-processing Platform which provides real-time, six-band, processing of microphone input from ear-level transducer assemblies. This system provides a more realistic participant experience, finer control of level and spectrum, and places no limits on the speech materials used for self-adjustment and outcome assessment. Ongoing work investigates the need for the speech-perception test as part of the self-adjustment protocol and the importance of the level and spectrum from which users make their initial adjustments.

2:20

1pPPa5. Tracking eye and head movements in natural conversational settings: Effects of hearing loss and background noise level. Hao Lu (Psych., Univ. of Minnesota, 75 East River Rd., Minneapolis, MN 55455, luxx0489@umn.edu), Martin McKinney, Tao Zhang (Starkey Hearing Technologies, Eden Prairie, MN), and Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN)

Although beam-forming algorithms for hearing aids can produce gains in target-to-masker, the wearer's head will not always be facing the target talker, potentially limiting the value of beam-forming in real-world environments, unless eye movements are also accounted for. The aim of this study was to determine the extent to which the head direction and eye gaze track the position of the talker in natural conversational settings. Three groups of participants were recruited: younger listeners, older listeners with clinically normal hearing, and older listeners with mild-to-moderate hearing loss. The experimental set-up included one participant at a time in conversation with two confederates approximately equally spaced around a small round table. Different levels of background noise were introduced by playing background sounds via loudspeakers that surrounded the participants in the conversation. In general, head movements tended to undershoot the position of the current talker, but head and eye movements together generally predicted the current talker position well. Preliminary data revealed no strong effects of hearing loss, or background noise level on the amount of time spent looking at the talker, although younger listeners tended to use their eyes, as opposed to head movements, more than the older listeners. [Work supported by Starkey Laboratories.]

2:40

1pPPa6. A visually guided beamformer to aid listening in complex acoustic environments. Todd R. Jennings and Gerald Kidd (Dept. of Speech, Lang. & Hearing Sci., Boston Univ., Boston University College of Health and Rehabilitation Sciences: Sargent College, 635 Commonwealth Ave., Boston, MA 02215, toddj@bu.edu)

The purpose of this talk is to provide an overview of recent work on the visually guided hearing aid (VGHA; Kidd *et al.* JASA **133**, EL202). The latest prototype VGHA consists of a lightweight, head-mounted, 18-microphone array combined with PC-based software to create a highly-tuned acoustic beamformer steered in real time by an integrated binocular eye tracker. Recent work from our group on the two main functional sub-components of the VGHA (beamforming and eye-gaze control) focused on the benefits that may be obtained by listeners with hearing loss (and in some cases by listeners with normal hearing) attempting to understand speech masked by one or more competing sounds (such as Gaussian noise or additional speakers). These benefits include improved signal-to-noise ratio under different masked conditions and faster steering (compared to fixed directional amplification steered by head turns). The VGHA may be simulated over headphones by PC-based implementations using HRTFs and an optional external eye-tracker. This has allowed testing a number of variations of beamformer properties and the evaluation of multiple types of sound processing strategies including the incorporation of natural binaural cues. Overall, the VGHA holds considerable promise for improving selective listening in a variety of complex listening conditions.

3:00–3:15 Break

3:15

1pPPa7. Robust and real-time decoding of selective auditory attention from M/EEG: A state-space modeling approach. Sina Miran (Dept. of Elec. and Comput. Eng., Univ. of Maryland, 2351 A. V. Williams Bldg., 8223 Paint Branch Dr., College Park, MD 20742), Sahar Akram (Facebook, Menlo Park, CA), Alireza Sheikhattar, Jonathan Z. Simon (Dept. of Elec. and Comput. Eng., Univ. of Maryland, College Park, MD), Tao Zhang (Starkey Hearing Technologies, Eden Prairie, MN), and Behtash Babadi (Dept. of Elec. and Comput. Eng., Univ. of Maryland, College Park, MD, behtash@umd.edu)

Humans are able to identify and track a target speaker amid a cacophony of acoustic interference, which is often referred to as the cocktail party phenomenon. Results from several decades of studying this phenomenon have culminated in recent years in various promising attempts to decode the attentional state of a listener in a competing-speaker environment from M/EEG recordings. Most existing approaches operate in an offline fashion and require the entire data duration and multiple trials to provide robust results. Therefore, they cannot be used in emerging applications such as smart hearing aids, where a single trial must be used in real-time to decode the attentional state. In this work, we close this gap by integrating various techniques from state-space modeling paradigm such as adaptive filtering, sparse estimation, and Expectation-Maximization, and devise a framework for robust and real-time decoding of the attentional state from M/EEG recordings. We validate the performance of this framework using comprehensive simulations as well as application to experimentally acquired M/EEG data. Our results reveal that the proposed real-time algorithms perform nearly as accurate as the existing state-of-the-art offline techniques, while providing a significant degree of adaptivity, statistical robustness, and computational savings.

3:35

1pPPa8. Towards cognitive control of hearing instruments using EEG measures of selective attention. Torsten Dau, Jonatan Maercher Roersted, Søren Fuglsang, and Jens Hjortkjær (Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, tdau@elektro.dtu.dk)

Single-trial EEG measures of selective auditory attention have recently suggested the perspective of decoding who a listener is focusing on in multi-talker scenarios. Here, we report results from work within the COCOHA (Cognitive Control of a Hearing Aid) project investigating the possibility of integrating EEG into neuro-steered hearing instruments. Our EEG decoding strategy relies on measuring cortical activity entrained to envelope fluctuations in the attended speech signal. Currently, a major challenge has been to obtain robust EEG measures of selective attention in older hearing-impaired (HI) listeners. We report our recent COCOHA attempts to decode selective attention from the EEG of hearing-impaired (HI) listeners. Aided HI listeners and age-matched normal-hearing controls were presented with competing talkers at 0 dB target-to-masker ratio and instructed to attend to one talker. We show that single-trial decoding accuracies similar to those reported for younger listeners can be obtained with both groups of older listeners (70–100% correct single-trial classification). Importantly, we did not find differences in decoding accuracies between the NH and the aided HI listeners. Although numerous other challenges involved in integrating EEG signals in hearing instruments are evident, our results suggest that single-trial attention decoding is possible with hearing impaired listeners.

3:55

1pPPa9. Towards transparent EEG: Ear-centered EEG acquisition using the cEEGrid. Martin G. Bleichner and Stefan Debener (Psych., Univ. of Oldenburg, Universitaet Oldenburg, Ammerlaender Heerstrasse, Oldenburg 26111, Germany, martin.bleichner@uol.de)

Ear-centered electroencephalography (EEG) allows for inconspicuous acquisition of EEG signals. Different approaches of placing electrodes in the ear canal, the concha, or around the ear have been proposed. As integral part of a hearing device, they promise to capture the neural correlates of a user's listening intent or listening effort in everyday situations. We have developed a ten electrode c-shaped array (cEEGrid); the flex-printed grid is positioned around the ear using an adhesive tape and allows for extended (>8 hours) high quality EEG recordings. We have shown that neural activity related to auditory processing originating from different neural sources can be reliably recorded. The signals recorded from different locations around the ear contain non-redundant information. The signal similarity between electrodes depends on their respective angle relative to the reference electrode. If this angle difference is small (20°) the signals recorded at these electrodes correlate highly (correlation score around 0.8). If this angle difference is large (160°) the signals recorded at these electrodes are uncorrelated. From this we conclude that the distributed electrode placement around the ear provides an interesting perspective on different neural processes. In combination with smartphone based experimentation, signal acquisition and analysis ear-centered EEG may soon enable the use of brain signals captured in daily life situations.

4:15

1pPPa10. Robust speech processing in human auditory cortex. Nima Mesgarani (Elec. Eng., Columbia Univ., 560 Riverside Dr. 17B, New York, NY 10027, nima@ee.columbia.edu)

The brain empowers humans with remarkable abilities to navigate their acoustic environment in highly degraded conditions. This seemingly trivial task for normal hearing listeners is extremely challenging for individuals with auditory pathway disorders, and has proven very difficult to model and implement algorithmically in machines. In this talk, I will present the result of an interdisciplinary research effort where invasive and non-invasive neural recordings from human auditory cortex are used to determine the representational and computational properties of robust speech processing in the human brain. These findings show that speech processing in the auditory cortex is dynamic and adaptive. These intrinsic properties allow a listener to filter out irrelevant sound sources, resulting in a reliable and robust means of communication. Furthermore, incorporating the functional properties of neural mechanisms in speech processing models greatly impact the current models of speech perception and at the same time, lead to human-like automatic speech processing technologies.

4:35

1pPPa11. An optimization model for electroencephalography-assisted binaural beamforming. Wenqiang Pu (Xidian Univ., Shenzhen, Guangdong, China), Jinjun Xiao, Tao Zhang (Starkey Hearing Technologies, Eden Prairie, MN), and Zhi-Quan Luo (Shenzhen Res. Inst. of Big Data, Longxiang Rd. 2001, Shenzhen, Guangdong, China, luoxx032@umn.edu)

In a multi-speakers noisy environment, the incoherent correlation between the electroencephalography (EEG) signal and the attended speech envelope provides a *user-informed* way to design the beamformer for preserving the attended speech and suppressing others. In this work, we exploit such incoherent correlation property in terms of Pearson correlation and propose a unified optimization model for simultaneously designing beamformer and aligning *attention preference* of the user to each speech source. To balance different design considerations, the proposed optimization formulation makes a trade-off among aligning attention preference, controlling speech distortion, and reducing noise in a weighted manner in the objective function. Specifically, the attention preference is aligned by maximizing the Pearson correlation between the envelope of beamforming output and the linearly transformed EEG signal. To control speech distortion with flexibility, the spatial response of the beamformer to each source is penalized in a min-max sense. And the noise is further reduced by minimizing its mean squares at beamforming output. Experiments on collected EEG signal in two speakers environment demonstrate the effectiveness of the proposed model.

4:55

1pPPa12. Realistic virtual audiovisual environments for evaluating hearing aids with measures related to movement behavior. Maartje M. Hendrikse, Gerard Llorach, Giso Grimm, and Volker Hohmann (Medizinische Physik and Cluster of Excellence Hearing4all, Universität Oldenburg, Carl-von-Ossietzky-Straße 9-11, Oldenburg 26129, Germany, maartje.hendrikse@uni-oldenburg.de)

With increased complexity of hearing device algorithms a strong interaction between motion behavior of the user and hearing device benefit is likely to be found. To be able to assess this interaction experimentally more realistic evaluation methods are required that mark a transition from conventional (audio-only) lab experiments to the field. In this presentation, we describe our methodology for acquiring ecologically valid behavioral data in realistic virtual audiovisual testing environments. The methods are based on tools to present interactive audiovisual environments while recording subject behavior with gaze and motion tracking systems. The results of a study that evaluated the effect of different types of visual information (e.g., video recordings vs. animated characters) on behavior and subjective user experience are presented. It was found that visual information can have a significant influence on behavior and that it is possible to systematically assess this. Furthermore, first results are presented of two studies that observed head and eye movement behavior: (1) in typical everyday listening situations that were replicated with virtual audiovisual environments in the lab (e.g., cafeteria) and (2) when visual cues were presented via a head-mounted display or projected onto a panoramic cylindrical screen in front of the subject.

5:15

1pPPa13. Multimodal signal processing and machine learning for hearing instruments. Tao Zhang and Martin McKinney (Signal Processing Research, Starkey Hearing Technologies, 8602 Zachman Circle, Eden Prairie, MN 55344, tzhang28@ieee.org)

With advances in wireless technology and sensor miniaturization, more and more non-audio sensors become available to and are being integrated into hearing instruments. These sensors help not only improve speech understanding and sound quality, enhance hearing usability and expand the hearing instruments' capabilities to health and wellness monitoring. However, the introduction of these sensors also present a new set of challenges to researchers and engineers. Compared with traditional audio sensors for hearing instruments, these new sensor inputs can come from different modalities and often have different scales and sampling frequencies. In some cases, they are not linear or synchronized to each other. In this presentation, we will review these challenges in details in the context of hearing instruments applications. Furthermore, we will demonstrate how multimodal signal processing and machine learning can be used to overcome these challenges and bring a greater degree of satisfactions to the end users. Finally, future directions in multimodal signal processing and machine learning research for hearing instruments will be discussed.

MONDAY AFTERNOON, 7 MAY 2018 NICOLLET A, 1:00 P.M. TO 5:00 P.M.

Session 1pPPb

Psychological and Physiological Acoustics: Spectral, Pitch, and Physiological Measurements (Poster Session)

Tess K. Koerner, Chair
University of Minnesota, 3330 SE Tibbetts St., Portland, OR 97202

Posters will be on display from 1:00 p.m. to 5:00 p.m. To give contributors in this session an opportunity to view other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

Contributed Papers

1pPPb1. Human dissimilarity ratings of musical instrument timbre: A computational meta-analysis. Etienne Thoret (Schulich School of Music, McGill Univ., 555 Rue Sherbrooke Ouest, Montreal, QC H3A 1E3, Canada, etienne.thoret@mcgill.ca), Baptiste Caramiaux (CNRS LRI UMR 8623, Univ. Paris-Sud, Paris, France), Philippe Depalle, and Stephen McAdams (Schulich School of Music, McGill Univ., Montreal, QC, Canada)

Musical instrument timbre has been intensively investigated through dissimilarity rating tasks. It is now well known that audio descriptors such as attack time and spectral centroid, among others, account well for the dimensions of the timbre spaces underlying these dissimilarity ratings. Nevertheless, it remains very difficult to reproduce these perceptual judgments from distances computed on acoustical representations such as the waveform or the spectrogram. Interestingly, biologically inspired representations based on spectrotemporal modulation spectra such as spectrotemporal receptive fields (STRF) have been shown to be well-suited to reproduce human dissimilarity ratings (Patil *et al.*, 2012). Here, we propose a meta-analysis of seven former studies on timbre spaces in light of these recently developed representations. We implemented a computational framework that optimizes the correlation between the perceptual results and distances

obtained from a set of different acoustic representations, in particular through the STRF. We observed that distances computed from spectrotemporal modulation representations provide the best correlation with the perceptual results across the seven timbre spaces. Finally, we highlighted the parts of the representations contributing the most to the correlation suggesting new insights into the underlying perceptual metrics. [Supported by Canada Research Chair, NSERC (RGPIN-2015-05208, RGPAS-478121-15), (RGPIN-262808-2012), and EU MSCf (Project MIM, H2020-MSCA-IF-2014, GA no. 659).]

1pPPb2. Neural coding of perceptual temporal asymmetry for sounds with rising vs. falling intensity envelopes in non-attentive and attentive listening conditions. Bing Cheng (English Dept. & Inst. for Lang., Cognition and Brain Sci., Xi'an Jiaotong Univ., 28 Xianning St. West, School of Foreign Studies, Xi'an, Shaanxi 710049, China, bch@mail.xjtu.edu.cn), Yang Zhang (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), Keita Tanaka (School of Sci. and Eng., Tokyo Denki Univ., Hatoyama, Saitama, Japan), Robert S. Schlauch (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), and Toshiaki Imada (Inst. for Learning and Brain Sci., Univ. of Washington, Seattle, WA)

Behavioral studies have shown that ramped sounds are judged to be louder and longer than damped sounds. Here we employed magnetoencephalography (MEG) to examine cortical processing of the perceptual temporal asymmetry. The participants were 6 normal-hearing right-handed male adults. The synthesized stimuli included three kinds, pure tone, piano note, and broadband noise with time-reversed intensity envelopes for the ramped vs. damped comparison. Each stimulus was 200 ms. In the non-attentive condition, subjects were instructed to watch a silent movie and ignore the randomly presented tones. In the attentive condition, listeners were required to judge whether the first or the second of paired sounds was longer. The stimuli were presented at 50 dB SL. The behavioral results replicated previous temporal asymmetry findings for all three types of stimuli. There were significant effects of stimulus type and stimulus order in the MEG ON and OFF responses. Despite the stimulus type effect, the ramped tonal stimuli consistently elicited smaller and later N1m responses than the damped controls in both ignore and attend conditions. These data supports distinct cortical mechanisms for coding the stimulus type information and subjective duration information of the ramped vs. damped auditory stimuli.

1pPPb3. Optimal frequency filtering of auscultation sounds. Lukasz Nowak (Inst. of Fundamental Technolog. Res., Polish Acad. of Sci., ul. Pawinskiego 5B, Warszawa 02-106, Poland, lnowak@ippt.pan.pl)

Most of the auscultation sounds do not reveal any significant single-frequency components, and their acoustic energy is concentrated in the low frequency region—up to about 100 Hz, falling even below the threshold of hearing. Such character is determined not only by the vibroacoustic behavior of sources, but mostly by high damping introduced by the sound transmission path through tissues underlying the skin surface. The contained diagnostic information is very subtle, and thus it can be easily masked by internal or external noise sources. Not all of those corrupting signals can be efficiently blocked, hence frequency filtering is the most obvious solution for improving the diagnostic capabilities. Many various filtering strategies and techniques were developed and implemented in both acoustic and electronic stethoscopes, however they are based primarily on (not always correct) intuition and subjective evaluation, without implementation of any accurate and objective measurement means or optimization algorithms. The present study introduces various signal to noise ratio (SNR) measures, applicable for different examination cases. Frequency filtering optimization strategies, for maximizing the values of the introduced coefficients for different heart and lung auscultation sounds, are presented.

1pPPb4. The effects of 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 key factor on correct phone recognition in Normal Hearing (NH) and Hearing Impaired (HI) listeners, is the intensity of primary cue. One can assess this intensity for a given speech sound, by examining it at various Signal to Noise Ratios (SNR) presented to NH listeners, and detect the threshold in which listeners recognized the token at least 90% correct (SNR_{90}). For each token, we have determined the time-frequency window corresponding to correct recognition, as well as the conflicting cues. Two sets of tokens T_1 and T_2 having same consonant-vowels but different talkers with distinct SNR_{90}s had been presented at flat gain (frequency independent) at listeners' most comfortable level (MCL). We studied the effects of frequency fine-tuning of the primary cue by presenting tokens of same consonant but different vowels with similar SNR_{90}s. Additionally, we investigated the role of changing the intensity of primary cue on HI phone recognition, by presenting tokens from both sets T_1 and T_2. This presentation discusses how the frequency an/or intensity of the primary cue changes the confusion pattern of phone recognition for HI listeners, given a flat gain condition at MCL. We will also explore the effect of these changes on conflicting cues.

1pPPb5. An oscillatory template pitch model. David A. Dahlbom and Jonas Braasch (Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, dahlbd@rpi.edu)

Traditional approaches to explaining missing fundamental and pitch-shift phenomena have relied on either the fitting of harmonic templates to spectral information (generally assumed to correspond to a neural excitation pattern), or by performing an autocorrelation-based calculation on temporal information. In this paper, an alternative approach relying on the dynamics of phase-locking oscillators is proposed. Rather than applying some decoding procedure, such as spectral analysis or autocorrelation, a pitch estimate is given by the state (firing frequency) of an appropriately excited oscillator in a template structure operating on peripheral information. This approach is shown to reproduce many of the classical pitch shift-phenomena. In addition to a direct implementation in terms of phase-locking oscillators, a simplified model consisting of arrays of adaptive templates, operating directly on timing information, is also presented. This latter approach offers a straightforward way to tune the model in accordance with existing psychoacoustical data and suggests an approach to modeling pitch strength and multiple f0 detection. The overall goal is to suggest the advantages of a modeling approach that relies on arrays of tuned, adaptive elements which are physiologically-inspired, though not intended to be detailed representations of precise physiological components. [Work was supported by NSF BCS-1539276.]

1pPPb6. Pitch synchronous speech analysis for the assessment of subjects with Parkinson's disease. Sai Bharadwaj Appakaya and Ravi Sankar (iCONS Res. Lab, Dept. of Elec. Eng., Univ. of South Florida, 4202 E Fowler Ave. ENB 381, Tampa, FL 33620, saibharadwaj@mail.usf.edu)

Analysis of speech samples from subjects with Parkinson's Disease (PD) is a field of growing research interest. Studies in this field predominantly include pitch-based features with the intuition that PD affects the movement of musculature involved in speech production. These features are usually extracted on a segment of fixed length that traverses over the entire speech sample. This methodology, however, gives the net estimate of the feature over each segment and cannot account for the fast variations that transpire before and after a vocal fold closure. In this paper, we present a pilot study with the focus on in-depth pitch synchronous analysis that can fill-in the gap by capturing the said variations. Speech samples from 22 patients are preprocessed and used for analysis. Data that can pin point the precise indices of the vocal fold closures for each cycle are extracted in preprocessing and features that can capture the swift variations are extracted from every cycle and used for analysis. The results show a good margin in features between PD and healthy speech that reinforced the intuition.

1pPPb7. Acoustic correlates of auditory object perception: Speakers, musical timbres, and environmental sounds. Mattson Ogg and L. Robert Slevc (Neurosci. and Cognit. Sci., Univ. of Maryland, College Park, Biology/Psych. Bldg. Rm. 3150, 4094 Campus Dr., College Park, MD 20742, mogg@umd.edu)

Human listeners must identify and orient themselves to auditory objects in their environment. What acoustic features support a listener's ability to differentiate the variety of sound sources they might encounter? Typical studies of auditory object perception obtain dissimilarity ratings between pairs of objects, often within a single category of sound. However, such an approach precludes an understanding of general acoustic features that might be used to differentiate sounds across categories. The present experiment takes a broader approach to the analysis of dissimilarity ratings by leveraging the acoustic variability within and between different sound categories as characterized by a large, diverse set of 36 sound tokens (12 speech utterances from different speakers, 12 instrument timbres, and 12 everyday objects from a typical human environment). We analyze multidimensional scaling results as well as models of trial- level dissimilarity ratings as a function of different acoustic representations including spectral, temporal and noise features as well as modulation power spectra and cochlear spectrograms. In addition to previously noted differences in spectral and temporal envelopes, results indicate that listener's dissimilarity ratings are also related to spectral variability and noise, particularly in differentiating sounds between categories. Dissimilarity ratings also appear to closely parallel sound identification performance.

1pPPb8. A pitch encoding model based on the intrinsic oscillation circuit: Effects of the time constant and input stimulus types. Minoru Tsuzaki (Kyoto City Univ. of Arts, 13-6 Kutsukake-cho, Oe, Nishikyo-ku, Kyoto 610-1197, Japan, minoru.tsuzaki@kcua.ac.jp) and Katuhiro Maki (Faculty of Human Informatics, Aichi Shukutoku Univ., Aichi, Japan)

Many temporal models for pitch perception have adopted a configuration of delay-lines and coincidence detectors after the cochlear filtering. Autocorrelation functions are a usual way of its implementation. However, a series of experiments by the authors' group have revealed that the perceived pitch would shift upwards by the effect of aging. Because the autocorrelation simply represents the time intervals statistics in the physical domain, the aging cannot affect this statistics. Therefore, a further pitch encoding process where the physical (physiological) temporal intervals are mapped against any internal reference in the brain. We propose a model comprised of bank of self-oscillatory circuits and the coincidence detectors. The periods of oscillations are intrinsic characteristics of the neural circuit. The proposed model could pick up the fundamental periods of a various types of stimuli, i.e., pure tones, missing fundamentals and iterated ripple noises. To check an effect of aging, the changes in the time constant of the temporal waveform of the oscillation was investigated. Although the shortest sensitive period increased due to the longer time constant, the age-induced pitch shift could not be predicted. It is necessary to assume a systematic lengthening of the period of each oscillation to predict the pitch shift.

1pPPb9. Speaker-dependent low-level acoustic feature extraction for emotion recognition. Tejal Udhan and Shonda Bernadin (Elec. Eng., Florida State Univ., 2525 Pottsdamer St., Tallahassee, FL 32310, tu13b@my.fsu.edu)

In this paper, accuracy for emotion recognition using low- level acoustic features is investigated. The aim of any speech emotion recognition system is to extract acoustic features that are representative of the emotional state of the speaker. Frequency formants, intensity, and pitch are the low-level features proposed for characterizing four different emotions, anger, happy, sadness, and neutral, using acoustic data. Low- level features describe the acoustic, prosodic, and spectral properties of the speech signal and limit the complexity of emotion recognition systems. An algorithm is designed for characterizing each emotion using the acoustic features. It has been proven that various aspects of a speaker's physical and emotional state can be identified by speech alone. However, the accuracy of such analyses has not been optimized due to acoustic variabilities such as length and complexity of human speech utterance, gender, speaking styles and speech rate. It has also been found that speaker-dependent systems are more accurate in emotion recognition than that of speaker-independent systems. Since speech emotion recognition is relatively a newer field, the set of most powerful features which can distinguish different emotions is not defined; hence, examining the accuracy of emotion recognition using selected acoustical features is an important task.

1pPPb10. Predicting timbral and perceptual characteristics of orchestral instrument combinations. Aurelien Antoine and Eduardo Miranda (Interdisciplinary Ctr. for Comput. Music Res. (ICCMR), Univ. of Plymouth, Plymouth University, The House Bldg. - Rm. 304, Plymouth PL4 8AA, United Kingdom, aurelien.antoine@postgrad.plymouth.ac.uk)

Orchestration is a compositional practice that consists of writing for several instruments. This process often involves harnessing each instrument's sound to create sonic textures that could not be achieved with a single instrument. These sound fusions are usually sought by composers to express specific perceptual effects. However, the number of potential combinations is significant. Testing and analyzing all combinations to identify the ones matching the desired perceptual effects is logistically and computationally complex. Using supervised learning methods to create regression and classification models, it is possible to predict specific timbral and perceptual characteristics from information about a combination of different orchestral instruments. Such developments would provide methods to estimate the perception of instrument timbre fusions directly from abstract information. Similar methods could potentially be applied to other types of sources and predict specific perceptual characteristics without the need to perform an acoustical and psychoacoustical analysis on every audio sources.

1pPPb11. Informational masking of calm speech targets by emotional speech maskers. Shae D. Morgan (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Ste. 1201, Salt Lake City, UT 84112, shae.morgan@utah.edu)

Recent research suggests that some types of emotional speech are more intelligible than others when presented in background noise (Dupuis & Pichora-Fuller, 2015). Yet, emotional speech can often be present as the background noise itself. Attentional mechanisms and auditory stream segregation capabilities likely impact target word recognition in these situations with emotional speech maskers. The present study examined calm target sentences in masking background combinations of four emotions (calm, sad, happy, and angry) and two masker types (2-talker babble and speech-shaped noise). The emotion categories differ from one another on perceptual dimensions of activation (low to high) and pleasantness (unpleasant to pleasant). Speech-shaped noise maskers were spectrally and temporally similar to the 2-talker babble maskers. Performance was compared between the two masker type conditions to quantify the amount of informational masking induced by different emotional speech maskers. Furthermore, the number of target-masker confusions were identified for each masker emotion and subsequently analyzed. Results showed lowest target recognition with calm speech maskers, consistent with auditory stream segregation framework; however, confusions and informational masking were high for angry and happy speech maskers, which suggests that auditory attention to high-arousal stimuli may have influenced performance.

1pPPb12. Mechanical sound similarity and categorization. Annamarie Rosen, Nirmal Kumar Srinivasan (Audiol., Speech-Lang. Pathol., and Deaf Studies, Towson Univ., 8000 York Rd., Towson, MD 21252, nsrinivasan@towson.edu), Jeremy Gaston, and Brandon Perelman (Human Res. and Eng. Directorate, U.S. Army Res. Lab., Adelphi, MD)

Recent studies suggest that perceptual similarity among sounds can be used to predict accuracy in perceptual tasks (Dickerson & Gaston, 2014; Gaston, *et al.*, 2017). We operationalize this similarity as distance in a 2D similarity space generated using multidimensional scaling (MDS) on pairwise similarity stimulus ratings. We tested listeners' ability to categorize mechanical environmental sounds, and modeled the relationship between stimulus categorization and similarity. Stimuli were 18 mechanical sounds selected hierarchically from three broad categories (vehicles, power tools, and household appliances), and three subcategories. Thirty-five normal

hearing listeners completed pair-wise similarity rating and categorization tasks. Perceptual similarity ratings from all paired stimuli were used to develop an MDS solution. Stimuli in the solution generally were segregated by broad category membership, with some exceptions. Categorization performance for each broad category varied, with vehicles being the easiest and household appliances being the most difficult to categorize. MDS distances among the stimuli were used in a simple regression model to predict performance in the categorization task and explain confusions in categorization. The results of this study indicate that perceptual similarity is a strong predictor of variance in listeners' judgments of complex environmental sounds.

1pPPb13. The effects of carrier bandwidth and intensity on spectral ripple perception in listeners with hearing loss. Evelyn E. Davies-Venn, Kristi Oeding, and Andrew Haug (SLHS, Univ. of Minnesota, 164 Pillsbury Dr SE, Minneapolis, MN 55445, venn@umn.edu)

Listeners with hearing loss often need to listen to speech signals at high intensities to ensure proper audibility. Spectral processing deteriorates at high intensities, but signal acoustics such as bandwidth and filtering can be used to mitigate this deterioration. This study evaluated the effect of carrier bandwidth on broadband spectral processing at high intensities for individuals with hearing loss. Spectral modulation detection thresholds were measured using a rippled noise carrier with varying bandwidths from 1 to 4 octaves, to assess whether individuals with hearing loss were more susceptible to the deleterious effects of high-intensity compared to their counterparts with normal hearing. Results show that spectral processing degrades at high intensities. Listeners with hearing loss benefited from the increase in signal bandwidth at high intensities much more than their counterparts with normal hearing. Our findings suggest that spectral processing for broadband signals involve within and across channel resolution and both succumb to the negative effect of level-induced broadening of auditory-filter bandwidths. A better understanding of how filtering can be used to alleviate the adverse effect of high-intensity signals could prove useful for improving outcomes for those individuals whose only viable treatment option often involves listening to speech signals at high intensities.

1pPPb14. Multiple mechanisms in pitch perception revealed by individual differences. Malinda J. McPherson (Speech and Hearing BioSci. and Technol., Harvard Univ., MIT Bldg. 46-4078, 43 Vassar St., 46-4078, Cambridge, MA 02139, malindamcpherson@g.harvard.edu) and Josh McDermott (Brain and Cognit. Sci., Massachusetts Inst. of Technol., Cambridge, MA)

Pitch perception is traditionally assumed to involve the estimation of fundamental frequency (F0) from sound. Some pitch-related tasks, such as those involving musical intervals, are more difficult with inharmonic sounds (lacking a clear F0). Other tasks, such as up-down pitch discrimination, are unaffected by inharmonicity, suggesting that listeners track the spectrum, not the F0. To test whether these effects reflect distinct mechanisms, we measured individual differences among large sets of participants on a battery of tasks. Performance on tasks using a common mechanism should be correlated across participants. Two tone up-down discrimination for wideband harmonic tones was highly correlated with performance for wideband inharmonic tones. To isolate F0-based pitch, we also measured discrimination for pairs of harmonic tones with non-overlapping sets of harmonics. Harmonic and inharmonic discrimination did not strongly correlate with this latter task. These results suggest that the same mechanism subserves up-down change detection for harmonic and inharmonic tones, and that this mechanism is distinct from F0-based pitch estimation. We also found weak correlations between spectral centroid discrimination and all other pitch tasks, suggesting separate mechanisms for fine and coarse spectral comparisons. The results suggest an additional pitch mechanism for tracking shifts in the spectrum rather than F0.

1pPPb15. The function of F0-based pitch. Malinda J. McPherson (Speech and Hearing BioSci. and Technol., Harvard Univ., MIT Bldg. 46-4078, 43 Vassar St., 46-4078, Cambridge, MA 02139, malindamcpherson@g.harvard.edu) and Josh McDermott (Brain and Cognit. Sci., Massachusetts Inst. of Technol., Cambridge, MA)

Pitch is traditionally described as the perceptual correlate of a sound's fundamental frequency (F0). The importance of perceiving F0 is often attributed to the need to compare sounds with different timbres, for which spectral comparisons might be error-prone. Alternatively, because the F0 is a summary measure of a set of harmonic frequencies, it could provide a compressed representation of sound for memory storage. To clarify the function of F0-based pitch, we tested its importance for discriminating sounds with different spectra, and sounds separated by delays, using both harmonic and inharmonic stimuli. If a task relies on F0-based pitch, performance should be impaired for inharmonic stimuli. We observed an advantage for harmonic stimuli for discrimination of synthetic tones with extreme (and unnatural) spectral variation. However, no such advantage was observed for instrument or speech fragments (with naturally occurring spectral variation). By contrast, discrimination was better for harmonic tones separated by brief (5–10 s) delays, suggesting that F0-based pitch aided storage across the delay. The results indicate that the spectral variation found in natural pitched sounds does not itself make listeners reliant on F0 estimation. The results raise the possibility that F0 primarily serves to facilitate compression of sounds for memory storage.

1pPPb16. Enhanced sine wave speech with an added cue for pitch improves tone perception in Cantonese. Amy Wu (Linguist, Brooklyn College, CUNY, Brooklyn College Linguist Program, 2308 Boylan Hall, 2900 Bedford Ave., Brooklyn, NY 11210, di5tortion@yahoo.com) and Jon Nissenbaum (Linguist, Brooklyn College and the Graduate Ctr., CUNY, Brooklyn, NY)

Cantonese has six lexical tones (four level and two rising) that distinguish otherwise identical syllables. Although it is known that other acoustic cues besides f0 (e.g., voice quality) enter into tone perception, it is unknown whether f0 alone provides a sufficient cue. To test this, modified Sinewave Speech (SWS) replicas of Cantonese sentences were presented to listeners. SWS replaces vocal tract formants with frequency-modulated sinusoids, and has been shown to support perception of phonemic content of speech. However, because formant trajectories alone lack information about f0 and voice quality, standard SWS is unsuitable for studying tone perception (Feng *et al*., 2012, JASA131/EL133; Rosen & Hui, 2015, JASA138). To create the stimuli, the lowest sinusoid of the SWS replica (representing F1) was replaced with a complex tone constructed using a bandpass with the same center frequency as F1, just wide enough at any timepoint for two harmonics of an independently specified f0 contour. This two-component tone implies a missing fundamental, allowing simultaneous impression of harmonic direction and F1 direction. Modified and unmodified SWS replicas of Cantonese words were presented to native Cantonese speakers to ascertain whether implied f0 improves tone perception; preliminary results support an effect of modified SWS.

1pPPb17. Modified two-component Shepard tones and their application to sine wave speech. Jon Nissenbaum (Linguist, Brooklyn College and the Graduate Ctr., CUNY, Dept. of English, 2308 Boylan Hall, Brooklyn College, 2900 Bedford Ave., Brooklyn, NY 11210, jnissenbaum@brooklyn.cuny.edu)

Sinewave Speech (SWS), consisting of several frequency-modulated sinusoids representing vocal tract formants, elicits perception of words/sentences, making it useful for studying the perceptual basis of speech. However, SWS contains no information relevant for pitch perception, making it a poor tool for investigating tone languages or prosody (Remez & Rubin, 1993, JASA94; Rosen & Hui, 2015, JASA138). The objective of this study was to develop a method for creating modified SWS replicas adding a minimal cue for pitch. The lowest sinusoid was replaced with a "modified Shepard tone:" a tone formed from a bandpass whose center frequency tracks F1, wide enough at any timepoint for exactly two harmonics of an independently specified f0 contour. It has been shown (Ritsma 1962, JASA34, Smoorenburg 1970, JASA48, Houtgast 1976, JASA60) that tones consisting

of 1–3 harmonics elicit perception of a missing fundamental. An initial experiment confirmed that a bandpass whose center frequency fell from 700 to 400 Hz over 400 ms, intersected with harmonics of a rising f0 (85–185 Hz), elicited a strong perception of a rising contour, and vice versa. A second experiment replaced F1 in a set of SWS replicas with these two-component tones and elicited robust perception of contrasting pitch contours.

1pPPb18. Short- and long-term memory for pitch and non-pitch contours in congenital amusia. Jackson Graves (Psych., Univ. of Minnesota, 75 E River Parkway, Minneapolis, MN 55455, grave276@umn.edu), Lesly Fornoni, Agathe Pralus (Lyon Neurosci. Res. Ctr. (CRNL), Université Lyon 1, Lyon, France), Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN), Anne Caclin, and Barbara Tillmann (Lyon Neurosci. Res. Ctr. (CRNL), Université Lyon 1, Lyon, France)

Congenital amusia is a disorder characterized by deficits in music and pitch perception, but the degree to which the deficit is specific to pitch remains unclear. Amusia often results in difficulties discriminating and remembering melodies and melodic contours. Non-amusic listeners can perceive contours in dimensions other than pitch, such as loudness and brightness, but it is unclear whether amusic pitch contour deficits also extend to these other auditory dimensions. This question was addressed by testing the identification of ten familiar French melodies and the discrimination of changes in the contour of novel four-note melodies. Melodic contours were defined by pitch, brightness, or loudness. Amusic participants were impaired relative to matched controls in all dimensions, but showed some ability to extract contours in all three dimensions. In the novel contour discrimination task, amusic participants exhibited less impairment for loudness-based melodies than for pitch- or brightness-based melodies, suggesting some specificity for pitch (including spectral pitch). Impairment in loudness contours for familiar melodies may have reflected the fact that the long-term memory representation of the familiar melody was defined in pitch. [Work supported by Erasmus Mundus Auditory Cognitive Neuroscience Network, NIH grant R01DC005216, LabEx CeLyA, and Cortex (ANR-11-IDEX-0007).]

1pPPb19. Frequency difference limens as a function of fundamental frequency and harmonic number. Anahita H. Mehta and Andrew J. Oxenham (Univ. of Minnesota, N640, Elliott Hall, 75 East River Parkway, Minneapolis, MN 55455, mehta@umn.edu)

Several studies have investigated the relation between the lowest harmonic present in a complex tone and fundamental frequency (F0) difference limens (F0DLs). It is generally assumed that F0DLs are smaller when lower harmonics are present and that the ability to discriminate small changes in F0 worsen as harmonic number increases. This worsening of performance has been attributed to a lack of peripherally resolved harmonics. This assumption was tested by measuring F0DLs for harmonic complexes where the lowest harmonic present in a twelve-harmonic complex tone varied from the 3rd to the 15th harmonic, with F0s varying from 30 Hz to 2000 Hz. The harmonics were presented in either sine or random phase and were embeded in threshold-equalizing noise. Aside from F0s between 100 and 400 Hz, performance did not follow the expected pattern of good performance with low-numbered (resolved) harmonics and poorer performance with high-numbered (unresolved) harmonics. At lower F0s, performance was relatively constant across the harmonic range; at higher F0s, performance generally degraded more than would be predicted based solely on harmonic resolvability. No clear effects of phase were observed. Overall, the results do not match the patterns expected from current models of complex pitch. [Work supported by NIH grant R01DC005216.]

1pPPb20. Gliding ripple speed thresholds for rippled spectra. Alexander Supin, Olga Milekhina, and Dmitry Nechaev (Institute of Ecology and Evolution, 33 Leninsky Prospect, Moscow 119071, Russian Federation, alex_supin@mail.ru)

Rippled noise is a productive model of natural signals with complex spectrum patterns. It was used as a test signal to measure spectrum-pattern resolution both in normal-hearing listeners and in hearing-impaired listeners and users of cochlear implants. However, a variety of natural auditory signals feature combined spectro-temporal patterns. These signals may be modeled by rippled noise with gliding ripples. In the present study, ripple gliding speed thresholds were measured in normal-hearing listeners. The highest gliding speed (expressed in oct/s or ripples/s) at which the gliding ripple pattern could be distinguished from a non-rippled noise was determined. The ripple gliding speed threshold expressed in ripple/s was inversely-proportional to the ripple density (ripples per octave, RPO); it decreased from approximately 400–500 ripple/s at RPO of 1/oct to approximately 50 ripple/s at RPO of 10/oct. Expressed in oct/s, the threshold decreased from 400 to 500 oct/s at RPO of 1/oct to 5 oct/s at RPO of 10/oct. Thus, the ripples fused into non-rippled spectrum at gliding speed more than 400 to 500 Hz/oct. At this speed, 400 to 500 ripples per second glide through an octave frequency band. [Work supported by The Russian Science Foundation, Grant 16-15-10046.]

1pPPb21. Masking of synthesized vowel sequences: The potential roles of perceptual segregation. Yi Shen and Dylan Pearson (Speech and Hearing Sci., Indiana Univ. Bloomington, 200 S Jordan Ave., Bloomington, IN 47405, shen2@indiana.edu)

Modulation masking is known to impact speech intelligibility, but it is not clear how this phenomenon interacts with the perceptual segregation between the target speech and background noise. Using an experimental task in which listeners identified synthesized-vowel sequences in amplitude-modulated noises, a previous study from our laboratory found few signs of modulation masking, contrary to the findings of speech-on-speech masking experiments. It is hypothesized that modulation masking at a low rate (<6 Hz) may reflect failures of segregating the target speech from the noise. The current study systematically removed segregation cues and evaluated the amount of modulation masking. These segregation cues included the periodicity in the temporal fine structure of the target vowel sequences (in Exp. I) and the temporal regularity of the vowel sequences (in Exp. II). Results will be discussed in terms of the potential interactions between modulation masking and perceptual segregation.

1pPPb22. Primacy and recency effects in serial recall of synthesized vowel sequences. Yi Shen (Speech and Hearing Sci., Indiana Univ. Bloomington, 200 S Jordan Ave, Bloomington, IN 47405, shen2@indiana.edu)

Recognizing running speech involves not only audibility of speech signals but also cognitive processes that support the encoding and retrieving of speech information to and from memory. The current study aims to separate the effects of audibility and cognitive load on speech recognition performance. Synthesized vowel sequences were presented in noise, and the listeners were instructed to recall all vowels in the sequences in the correct order. In separate conditions, the sequence lengths were 2, 5, and 8 vowels, the presentation rates were 1, 2, 4, and 6 Hz, and the vowel durations were between 12.5 and 100 ms. The signal-to-noise ratios used in the experiment were adjusted for each listener and each of the vowel durations so that the expected recognition performance for isolated vowels was always 80%. A primacy effect was observed and it was more prominent at higher presentation rates and longer sequence lengths. A recency effect was also observed in most listeners, but no significant effect of presentation rate was found.

1pPPb23. Late negative response revealed language-specific processing of word-level prosodic phonology in the adult brain. Luodi Yu, Yang Zhang (Dept. of Speech-Language-Hearing, Univ. of Minnesota, 164 Pillsbury Dr SE, Minneapolis, MN 55455, zhang470@umn.edu), Suiping Wang, and Jiajing Zeng (School of Psych., South China Normal Univ., Guangzhou, China)

Native speakers develop specialized neural architectures in the processing of auditory patterns found in their language. However, little is known about the neural basis of language-specific processing of phonological patterns beyond the syllable level. The current event-related potential (ERP) study investigated cortical processing of native vs. nonnative prosodic phonology in a disyllabic context. The participants were 18 normal native Chinese-speaking adults who had at least six years of English-as-a-second-language course work in school. The speech stimuli included disyllabic nonsense words spoken in Chinese and in English, which were matched in syllabic structure and digitally edited for matching in overall duration and intensity. The control nonspeech stimuli were hummed versions of the speech stimuli, which preserved the acoustic prosodic features but not the phonetic content. The ERP data were obtained with a passive listening paradigm, showing an enhanced late negative response (LNR) for the native vs. nonnative comparison in the speech condition but not in the nonspeech condition. These results provided the initial evidence that the LNR component can reflect language-specific differences in prosodic phonology beyond auditory processing of the critical acoustic cues at the supra-syllabic level.

1pPPb24. Analysis of the effect of different patologies on the sound transmission through the middle ear by means of a finite-element model. Lucas C. Lobato (Lab. of Acoust. and Vibrations, Federal Univ. of Santa Catarina, Av. Roraima n° 1000, Santa Maria, Rio Grande do Sul 97105-900, Brazil, lucascostalobato@gmail.com), Stephan Paul (Lab. of Acoust. and Vibrations, Federal Univ. of Santa Catarina, Joinville, Brazil), Julio A. Cordioli (Lab. of Acoust. and Vibrations, Federal Univ. of Santa Catarina, Florianópolis, SC, Brazil), and Evandro Maccarini (Lab. of Acoust. and Vibrations, Federal Univ. of Santa Catarina, Florianopolis, Brazil)

The middle ear is part of the human auditory system and its function is to transmit acoustical energy to the inner ear by means of mechanical vibrations. Finite-Element (FE) models of the human middle ear have been studied since 90s, making it possible to evaluate its dynamic behavior in different ways. In this work, an accurate FE model of the human middle ear is described and used for assessing the relationship between the reduction of sound transmission through the ossicular chain and hearing loss due to three different pathologies: tympanic membrane perforation, stapedial tendon ossification and stapes fracture. For these three pathologies, case studies from the literature with experimental hearing loss data are reviewed and clinical results are compared to the FE model predictions. In all cases a good relationship between sound transmission reduction predicted by the FE model and the hearing loss measured through audiometric tests were found. Results show that an accurate FE model of the human middle ear can be a powerful tool for clinical applications and research in otology, allowing to predict the effect of middle ear pathologies and to evaluate the results of corrective surgeries on hearing.

1pPPb25. Human inference of force from impact sounds: Perceptual evidence for inverse physics. James Traer and Josh McDermott (Brain and Cognit. Sci., MIT, 77 Massachusetts Ave., Cambridge, MA 02139, jtraer@mit.edu)

An impact sound is determined both by material properties of the objects involved (e.g., mass, density, shape, and rigidity) and by the force of the collision. Human listeners can typically estimate the force of an impact as well as the material which has been struck. To investigate the underlying auditory mechanisms we played listeners audio recordings of two boards being struck and measured their ability to identify the board struck with more force. Listeners significantly outperformed models based on simple acoustic features (e.g., signal power or spectral centroid). We repeated the experiment with synthetic sounds generated from simulated object resonant modes and simulated contact forces derived from a spring model. Listeners could not distinguish synthetic from real recordings and successfully estimated simulated impact force. When the synthetic modes were altered (e.g., to simulate a harder material) listeners altered their judgments of both material and impact force, consistent with the physical implications of the alteration. The results suggest that humans use resonant modes to infer object material, and use this knowledge to estimate the impact force, explaining away material contributions to the sound.

1pPPb26. A framework for evaluating perceptual interactions of various dimensions of sound for data sonifications. Samuel Chabot and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., 40 3rd St, Troy, NY 12180, chabos2@rpi.edu)

Due to its multidimensionality, sound is a powerful medium for presenting data. Various parameters, including pitch, loudness, timbre, rhythm, and spatial location, may be utilized to convey information. Interactions between dimensions can affect perceptions of this information. Experiments evaluating these interactions in the context of sonifications are important in designing more accurate auditory displays. Oftentimes, these experiments focus on a single pair of parameters. This project offers a framework for utilizing different combinations of parameters to streamline evaluations. A user interface allows for the direct manipulation, and subsequent juxtaposition, of these combinations. [Research funded by the Cognitive and Immersive Systems Laboratory at Rensselaer.]

1pPPb27. A study on the reducing stress of music listening by measuring heart rate. Bong Young Kim and myungjin bae (Sori Sound Eng. Lab, Soongsil Univ., 21-1, Garak-ro 23-gil, Songpa-gu, #203, Seoul 05669, South Korea, bykim8@ssu.ac.kr)

Humans are subject to many types of stress. Stress affects various nervous system transit hormone systems and weakens the immune system and weakens the autonomic nervous system, which slows the activities of various organs. It is already well known that those who overcome stress well have a strong immune system and overcome disease. It also achieves many successful results. This fact indicates how important it is to reduce stress. Music listening is one of the great ways to reduce stress. Music listening is known to stimulate the autonomic nervous system and the motor center to reduce complaints, edgy, and tension. Music listening also has an effect on mental treatment such as post-traumatic stress. In this paper, we tried to find out whether music listening helps to reduce stress through HRV (Heart Rate Variability) SDNN (Standard Deviation N-N Interval) obtained from heart rate measurement. Five subjects measured the heart rate by listening to four experiment song. The SDNN value increased when four of five people listened to all the experiment song. In the remaining one, the SDNN value was lowered for only one experiment song, and for the remaining three songs, the SDNN value was increased. However, the degree of reducing stress was different according to the musical preference of each individual. The results of this experiment showed that listening to music according to taste helps to reduce stress.

1pPPb28. Perceptual consequences of cochlear synaptopathy in middle age. Brooke Flesher, Alexandra Mai, Kelsey Dougherty, Jennifer M. Simpson, Michael G. Heinz, and Hari M. Bharadwaj (Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47906, bflesher@purdue.edu)

It is known that as the body ages, listening in noisy situations becomes increasingly more difficult. Middle-aged people have more difficulty than younger people in complex listening environments, even with similar audiometric thresholds. One hypothesis for why this might be is age-related cochlear synaptopathy. Cochlear synaptopathy has been robustly observed to occur with normal aging in animals, and in post-mortem studies of human temporal bones harvested at different ages. However, there are currently no clinical assessments to measure cochlear synaptopathy or the perceptual consequences of this phenomenon. Currently, we are evaluating a battery of clinical and laboratory measures by comparing young and middle-aged listeners with matched audiometric thresholds within the normal range. The battery includes both objective physiological measures that correlate with synaptopathy in animal models, and perceptual measures that hypothetically may be sensitive to synaptopathy-mediated loss of redundancy in afferent coding. Here, we discuss our initial findings within these parameters.

1pPPb29. Testing for evidence of cochlear synaptopathy in normal-hearing young adults with varying noise exposure history. Brandon T. Paul (Evaluative Clinical Sci., Sunnybrook Res. Inst., 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, brandon.paul@sunnybrook.ca), Natalie Tran, Sajal Waheed, Larry E. Roberts (Psych., Neurosci., & Behaviour, McMaster Univ., Hamilton, ON, Canada), and Ian C. Bruce (Elec. & Comput. Eng., McMaster Univ., Hamilton, ON, Canada)

Loss of high-threshold cochlear synapses following mild noise exposure is suggested to degrade amplitude modulation (AM) encoding in cases where hearing thresholds are normal. However, the relationship of AM encoding to noise exposure history has not been consistent in previous studies. We investigated this relationship in young adults with normal audiograms in two studies using different methods. Study 1 (N = 25) measured

the ~80 Hz electrophysiological envelope following response (EFR) and behavioural AM detection thresholds. Both measures were taken in quiet and in a narrowband background noise designed to attenuate low-threshold synapse contributions. When subjects were divided into two groups based on their noise exposure history, subjects with more noise exposure had smaller EFRs ($p = 0.0198$). AM detection was also poorer in these subjects, but this difference fell short of significance ($p = 0.067$). Study 2 (ongoing) also measures the EFR but employs an additional wider band of background noise to attenuate possible off-frequency contributions of low threshold fibers to AM coding. In addition, AM discrimination is tested instead of AM detection. The question is whether these modifications will reveal a more robust effect of noise exposure history on AM coding than did Study 1. [Work supported by NSERC of Canada.]

1pPPb30. Brain potentials to speech perception in noise for patients with schizophrenia. Lin Mi (Dept. of Psychiatry, The Affiliated Brain Hospital of Guangzhou Medical University (Guangzhou Huiai Hospital), 36 Mingxin Rd., Liwan District, Guangzhou, China, linmi83@hotmail.com), Le Wang, Shenglin She, Haijing Li (The Affiliated Brain Hospital of Guangzhou Medical Univ. (Guangzhou Huiai Hospital), Guangzhou, China), Chang Liu (Dept. of Commun. Sci. and Disord., the Univ. of Texas at Austin, Austin, TX), and Yingjun Zheng (The Affiliated Brain Hospital of Guangzhou Medical Univ. (Guangzhou Huiai Hospital), Guangzhou, China)

Schizophrenia is characterized with a series of cognitive impairment, including basic information processing such as auditory and speech perception. Especially, patients with schizophrenia experienced additional challenge for speech perception in noisy environment. In this study, brain activity in response to speech sounds was measured in quiet and 6-talker babble conditions from 25 schizophrenia patients and 20 healthy subjects. A double-oddball paradigm was employed, in which /ba/ was played as the standard stimulus with /da/ and /bu/ as the deviant stimuli. Results showed that the schizophrenia group had lower amplitude and longer latency of mismatched negativity (MMN) compared with control group in both the quiet and babble conditions. In addition, there was a significant interaction effect between listening condition and listener group, i.e., the 6-talker babble exerted greater disturbance for schizophrenia. Moreover, the analysis of theta band oscillation in the time window of MMN also supported the above results. These findings indicated that the deficits in the pre-attentive automatic auditory processing in the patients with schizophrenia may underpin their speech perception difficulty in the daily noise environment.

1pPPb31. Pupillometry as a measure of auditory cognitive processes and listening effort. Damian Almaraz (Psych., St. Olaf College, Northfield, MN), Brock Carlson (Neurosci., St. Olaf College, Northfield, MN), Elaine Grafelman, Jacob Ingalls, Yadi Quintanilla (Psych., St. Olaf College, 1520 St. Olaf Ave., Northfield, MN 55057, grafel1@stolaf.edu), Hieu-Trung Vu (Neurosci., St. Olaf College, Northfield, MN), and Jeremy Loebach (Psych., St. Olaf College, Northfield, MN)

Although many people with hearing impairments can accurately recognize words and sounds with proper treatments (hearing aids or cochlear implants), they often report fatigue and stress from doing so. These individuals need to exert more auditory cognitive processing effort to hear well, especially in adverse listening environments or when engaging in competing tasks. We hypothesize that the allocation of these auditory cognitive resources, or auditory cognitive "X-Factors," can explain variability in outcome across cochlear implant users. To find the "X-Factors," we use pupillometry as a physiological measure of cognitive load across auditory cognitive domains including short term memory (forward digit span) working memory (backward digit span), cognitive load (Paced Serial Addition Task), listening effort (speech in noise tests) and personality traits (motivation, self-efficacy, and adaptability). A custom-built eye tracking system using Pupil Labs software provides a cost-effective, dynamic, and fully mobile system to collect data in real time. Initial data from aging adults indicate significant correlations between cognitive abilities, pupil dilation, response strategy, and auditory performance. Understanding the "X-Factors" will help inform training and rehabilitation for hearing impaired individuals and cochlear implant users.

1pPPb32. Importance of time-frequency units for sentence recognition as a function of signal-to-noise ratio. Frederic Apoux (Otolaryngol. - Head & Neck Surgery, The Ohio State Univ. - Wexner Medical Ctr., 915 Olentangy River Rd., Columbus, OH 43212, fred.apoux@gmail.com), Brittney Carter (Speech & Hearing Sci., The Ohio State Univ., Columbus, OH), Karl P. Velik (Otolaryngol. - Head & Neck Surgery, The Ohio State Univ. - Wexner Medical Ctr., Columbus, OH), and Eric Healy (Speech & Hearing Sci., The Ohio State Univ., Columbus, OH)

Speech recognition in noise may be viewed as a classification problem, in which the auditory system selects a subset of time-frequency (T-F) units to build a representation of the signal. The present study introduces local signal-to-noise ratio (SNR) importance functions, an approach to determine the contribution to speech intelligibility of T-F units as a function of their SNR. Consistent with previous work from this group on auditory-channel independence, it was hypothesized that T-F units with a relatively equal mixture of signal and noise (around 0 dB SNR) may be the most disruptive for intelligibility because they should be the most difficult to classify. This hypothesis was assessed by measuring the intelligibility of sentences presented in a competing-speech background while discarding a fixed proportion of units all with the same local SNR. The stimuli were either vocoded or left unprocessed to evaluate the influence of fine-structure cues on classification and also help better understand why cochlear implant (CI) users experience greater difficulties in noise. Results were consistent with the ambiguity hypothesis. More importantly, they suggest that discarding the noisiest units may not be the most advantageous way to improve speech intelligibility in noise by CI users.

1pPPb33. The effect of masker type and masker timing on the serial recall of speech. Elin Roverud, Virginia Best, Christopher Conroy, and Gerald Kidd (Dept of Speech, Lang. & Hearing Sci., Bosto Univ., 635 Commonwealth Ave., Boston, MA 02215, emroverud@gmail.com)

Understanding speech in noise involves a complex set of processes including segregation of sources, selection of the target talker, and recognition and comprehension of the ongoing message. Previous studies often have focused on the segregation, selection and recognition processes by measuring the intelligibility of short strings of words presented concurrently with different types of maskers. However, these studies may not fully capture the effects of masking on all aspects of ongoing speech processing. For instance, masking can cause poorer recall for *sequences* of words even if a noise/competitor leads to near-perfect recognition of the individual items [e.g., Cousins *et al.*, 2014, Mem Cogn (42)]. Furthermore, in contrast to the typical design of speech recognition tasks, background noise usually does not cease during response to the message. In the present study, we examined listeners' serial recall of target speech with different maskers presented at different points during the trial. Speech-shaped noise or confusable speech maskers were presented colocated with or spatially separated from the target. The maskers were presented during target presentation, during the response interval, or continuously throughout the trial. Results will compare the influence of different types of masking during encoding and retrieval/response stages of processing.

1pPPb34. Estimating the duration of a refractory period in perceptual learning. Alex E. Clain (Commun. Sci. and Disord., Northwestern Univ., 5445 N Sheridan Rd., Apt. 2207, Chicago, IL 60640, alexanderclain2022@u.northwestern.edu) and Beverly Wright (Commun. Sci. And Disord., Northwestern Univ., Evanston, IL)

Auditory skills improve with practice. Producing across-day learning requires a sufficient number of trials per session; additional trials in the same session produce no additional improvement. Thus, trials appear to integrate within a session to reach a learning threshold, after which there is a refractory period during which trials do not contribute to learning. The refractory period duration must be <24 hours because training on consecutive days yields learning each day, but its exact duration is unknown. To narrow the range of possible durations, we trained young-adult, normal-hearing listeners on a basic auditory task (interaural-level-difference discrimination) either for two sessions of sufficient training separated by 30 minutes ($n = 10$) or 10 hours ($n = 10$), or for just one session ($n = 15$), and assessed their learning on the next day. The 30-minute group learned no more than the single-session group, indicating the second session did not

aid learning, whereas the 10-hour group learned more than the other two groups, indicating the second session provided additional benefit. These results suggest that the refractory period lasts >30 minutes, but <10 hours, and does not require sleep to reset. Knowledge of the refractory period duration will improve the efficiency of perceptual training for typical and clinical populations.

MONDAY AFTERNOON, 7 MAY 2018 GREENWAY C, 1:00 P.M. TO 3:00 P.M.

Session 1pSA

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

Benjamin Shafer, Chair

Technical Services, PABCO Gypsum, 3905 N 10th St, Tacoma, WA 98406

Contributed Papers

1:00

1pSA1. Characterizing stress corrosion cracking in stainless steel using nonlinear resonant ultrasound spectroscopy. Stephen M. Hogg and Brian E. Anderson (BYU Dept. of Phys. & Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602, jr.hoggcube@gmail.com)

During disposal, spent nuclear fuel rods are placed into stainless steel containers and may be stored this way for decades. In order to ensure there is as little radiation leakage as possible, the structure of the stainless steel needs to be evaluated and possible damage detected. In particular, one type of damage these containers are particularly susceptible to is stress corrosion cracking (SCC). Traditional (linear) evaluation methods can detect open cracks but often a closed portion of the crack extends further, compromising the air tight seal sooner. To this end, it has been proposed to use Nonlinear Resonant Ultrasound Spectroscopy (NRUS) to quantify the degree of cracking in a structure. In order to prove this concept, cylindrical 304L stainless steel rods were immersed in a heated 42% $MgCl_2$ solution to induce SCC in an accelerated manner to simulate the damage that is likely to occur on the actual containers. The rods were removed after different lengths of exposure. NRUS measurements were then conducted for longitudinal modes in the rods.

1:15

1pSA2. Acoustic reflection and transmission by sub-critical and super-critical elastic partitions with structural discontinuities forced by multiple-angle oblique broadband sound waves. Mauricio Villa, Donald B. Bliss, and Linda P. Franzoni (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., Box 90300 Hudson Hall, Durham, NC 27708, mauricio.villa@duke.edu)

An analysis is presented for acoustic reflection and transmission from an infinite fluid-loaded flexible barrier with spatially periodic discontinuities. The plate, with similar or dissimilar acoustic fluids on both sides, is excited by an oblique wave incident acoustic field. The fully-coupled structural/acoustic problem is treated by Analytical-Numerical Matching (ANM) which improves the numerical accuracy and convergence rate. Periodic spatial discontinuities, modelled by various boundary conditions, create deviations from specular directivity. For super-critical structural wave speeds, the scattering is characterized by a distribution of reflection and transmission angles centered around the Mach Angles. For sub-critical structural waves, energy redistribution is most pronounced at the structural resonances. These results are compared to a baffled finite barrier analyzed by approximate means. For the subcritical finite barrier, a fluid loading correction is introduced to the structural wavenumber to approximate the effects of the fluid coupling and radiation damping. For subsonic flexural waves, the radiation is analyzed in terms of the wavenumber transform and interpreted as a series of edge singularities dependent on the boundary conditions. For the super-critical configuration, the effects of the fluid loading are introduced by approximating the structural wavenumber from the dispersion relation of the free response of the unbounded structure.

1:30

1pSA3. Remote acoustic detection of localized delamination in vibrating composite layered 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)

Adhesively bonded, layered composite materials are ubiquitous in naval, aerospace, and automotive applications. One common challenge with such materials is the presence of delamination discontinuities at the interlaminar bonds which can have severe adverse effects to structural health and system performance. Worse yet, delamination is often visually indiscernible as it occurs below visually-obstructing surface layers, making non-visual defect detection techniques attractive in comparison. In this presentation, remote acoustic detection of localized delamination in a 0.3-m-square composite laminate plate subject to 100-6000 Hz noisy broadband forcing will be investigated. Remote measurements of the radiated acoustic field from composite plates with and without known localized delamination are collected with a 15-element microphone array. Here, statistical comparisons of spectral amplitudes between plates with and without damage are then used to evaluate remote acoustic detection performance. Additional signal processing steps are applied to the raw array data to improve performance robustness in the reverberant laboratory test setting. Considerations for detection performance include minimum detectable size of delamination, robustness to receiving-array location, and sensitivity to ambient noise. [Sponsored by NAVSEA through the NEEC and by the US DoD through an NDSEG Fellowship.]

1:45

1pSA4. Acoustic methods for regionalizing an impact force acting on a structure. W. S. Shepard and Jacob Davis (Mech. Eng., The Univ. of Alabama, Box 870276, Tuscaloosa, AL 35487, jadavis15@crimson.ua.edu)

It is often desired to know the location and magnitude of a force acting on a structure. Unfortunately, it is not always possible to install a sensor at the force location, particularly when that location is unknown. A structure subjected to an impact has many different vibrational modes that are excited to different levels based on the excitation location. These vibrations decay with time depending on their different rates of modal damping and the associated acoustic radiation characteristics. In this work, acoustic signatures for

a number of structure impact locations were measured, normalized relative to the force magnitude, and processed to examine the ability to correlate the acoustic signal to the force impact location. Various processing techniques, such as the Short Time Fourier Transform, were considered. A primary interest focused on the ability of single-number metrics to aid in identifying the force location. For the experiments, a football helmet structure was used and multiple impact locations were tested. The ability of these acoustic signatures, including those processed into single-number metrics, to aid in identifying the impact location was assessed. The location resolution for which these methods were effective was also evaluated.

2:00

1pSA5. A study on the oxygen generation in water molecules using sound wave. Yi Eun Young, Uk-Jin Song, and myungjin bae (Information and TeleCommun., Soongsil Univ., Soongsil Univ., Sangdo 1-dong, Dongjak-gu, Seoul 156-743, South Korea, go6051@naver.com)

Life is living in a different place. There are mammals and birds living in the air through oxygen. There is a life extension using oxygen present in the water. There are fish and mammals in the water that use oxygen in the water. The whales, belonging to the mammals in the sea, breathe out of the water at regular intervals and then back into the water. Fish do not hinder the extension of life without these actions. This study uses speakers as means for generating oxygen in water molecules. It is a study to decompose oxygen in water by using the dense wave of the speaker sound wave. It is a study that removes oxygen by affecting the bonding structure of water molecules through the vibration of sound waves. Unlike other chemical methods, this characteristic can generate oxygen without generating hydrogen.

2:15

1pSA6. Inter-digital transducers activated acoustic streaming in viscous liquid. Chuyi Chen, Peiran Zhang (Mech. Eng. and Mater. Sci., Duke Univ., 144 Hudson Hall, Box 90300, Durham, NC 27708, cc507@duke.edu), Zhangming Mao (Dept. of Eng. Sci. and Mech., The Penn State Univ., Durham, North Carolina), Yuyang Gu (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC), Francesco Costanzo (Dept. of Eng. Sci. and Mech., The Penn State Univ., University Park, PA), and Tony Jun Huang (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC)

Acoustic streaming induced by surface acoustic waves (SAWs) has been widely utilized in various medical and biology applications. For most of the applications, the microfluidic channel is attached to a piezoelectric substrate where the inter-digital transducers (IDTs) are fabricated on it to activate acoustic waves. The acoustic waves would propagate along the substrate surface and generate the "leaky wave" in the channel which causes the acoustic streaming. However, few studies are conducted to analyze the vibration mode of the substrate in the IDTs area and the following phenomenon (e.g., acoustic streaming generated by substrate vibration). In this study, we demonstrate a novel method which includes both simulation and experiment results to identify the substrate vibration and corresponding streaming while the liquid is loading upon the IDTs directly. With this method, we show that the acoustic streaming on top of the IDTs area has a "stagnation point" with a velocity potential to trap and manipulate the objects floating on the surface of the liquid, which is promising in many contactless biomedical applications.

2:30

1pSA7. Nonlinear resonant ultrasound spectroscopy using electromagnetic excitation. Joshua F. Gregg and Brian E. Anderson (Dept. of Phys. & Astron., Brigham Young Univ., BYU Dept. of Phys. and Astron., 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 the sample is typically done with a piezoelectric transducer. We have been applying an electromagnetic excitation developed for resonant ultrasound spectroscopy [S. Garrett, J. Acoust. Soc. Am. **88**(1), 210–221 (1990)] as an alternative for NRUS excitation. It involves gluing a coil of wire onto the end of a rod-shaped sample and placing it in a magnetic field. By controlling which part of the coil is in the strongest part of the magnetic field, we can control the principle direction of the driven oscillations in the rod. This method allows us to selectively excite longitudinal, torsional, and bending vibrations and measure the nonlinear properties of the sample for each type of vibration. We have applied this electromagnetic method to measuring the nonlinear properties of Berea sandstone, and we plan to use it to detect stress corrosion cracking in stainless steel samples. The purpose of this presentation is to illustrate the usefulness of the electromagnetic method.

2:45

1pSA8. Thermoacoustic instability in solid media. Haitian Hao, Carlo Scalo (Mech. Eng., Purdue Univ., Herrick Labs, 177 S. Russell St, Rm. 1007, West Lafayette, IN 47906, haoh@purdue.edu), Mihir Sen (Aerosp. and Mech. Eng., Univ. of Notre Dame, Notre Dame, IN), and Fabio Semperlotti (Mech. Eng., Purdue Univ., West Lafayette, IN)

The exploration of thermoacoustic oscillations can be traced back to the 19th century when Sondhauss (1850) performed an experimental investigation that marked the birth of modern thermoacoustic research. Since then, several practical applications of the basic thermoacoustic effect were explored and ultimately led to the design of engineering devices such as heat engines or refrigerators. This fascinating mechanism was believed to occur only in fluids while our study shows theoretical and numerical evidence of the existence of thermoacoustic oscillations also in solid media. Although this mechanism shares common aspects with the more familiar thermoacoustics of fluids, our analysis shows that the underlying physical mechanism presents some important and non-trivial differences with the traditional theory. The discovery and validation of this mechanism may lead to new ideas enabling the development of the next generation of ultra-compact and robust solid-state thermoacoustic devices for direct thermal-to-mechanical energy conversion, and vice versa.

Session 1pSCa

Speech Communication: South Asian Languages I

Kelly Berkson, Cochair
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Linguistics, Macalester College, 1600 Grand Ave, St. Paul, MN 55105

Indranil Dutta, Cochair
Department of Computational Linguistics, English and Foreign Languages University, Tarnaka, Osmania University Campus, Hyderabad 500605, India

Chair's Introduction—1:00

Contributed Papers

1:05

1pSCa1. Acoustic characteristics and perception of voiced and voiceless nasals in Mizo and Angami. Pamir Gogoi (Dept. of Linguist, Univ. of Florida, Gainesville, FL 32611, pgogoi@ufl.edu)

This study compares and contrasts acoustic characteristics (e.g., nasal murmur's formant and anti-formant frequencies, vowel formant frequencies, bandwidths, spectral tilts, etc.) of voiced and voiceless nasals in Mizo and Angami, Tibeto-Burman languages spoken in North-East India. Mizo contrasts voiced and voiceless nasals at the bilabial, alveolar, and velar places of articulation, whereas Angami contrasts them at four different places of articulation: bilabial, alveolar, velar, and palatal. Similar to voiceless nasals previously described for Burmese (Danstuji,1984; Bhaskararao & Ladefoged,1991), in Mizo, voiceless nasal murmur is followed by a period of voiced nasal murmur as it transitions into the following vowel, while it remains voiceless throughout in Angami. Results of a perception experiment also suggests that the voiced nasal murmur portion in Mizo voiceless nasals aided the identification of its place of articulation, but a heavier perceptual load is placed on the following vowel for place identification of voiceless nasals in Angami. **References:** Dantsuji, M. (1984). "A study on voiceless nasals in Burmese." Bhaskararao, P., & Ladefoged, P. (1991). "Two types of voiceless nasals," *J. Int. Phonet. Assoc.*, ***21***(2), 80–88.

1:20

1pSCa2. Stop laryngeal contrasts of endangered languages of Northern Pakistan. Qandeel Hussain and Jeff Mielke (Dept. of English (Linguist Program), North Carolina State Univ., 221 Tompkins Hall, Campus Box 8105, Raleigh, NC 27695-0001, qhussai@ncsu.edu)

Iranian, Indo-Aryan (Dardic), and Tibeto-Burman languages of Northern Pakistan are characterized by 2-, 3-, and 4-way laryngeal contrasts at four or five places of articulation (e.g., Balti, Kalasha, Wakhi, etc.). However, the number of stop laryngeal contrasts, their phonetic realization, and the exact place of articulation of retroflex stops (apical or sub-apical) are still controversial in most of the endangered languages of Northern Pakistan. The current study provides the first detailed acoustic and articulatory (Ultrasound for coronals and velars and aerodynamic for all stops) description of the stop laryngeal contrasts of six languages of Northern Pakistan: Balti (Tibeto-Burman), Burushaski (Isolate), Kalasha, Khowar, Shina (Indo-Aryan, Dardic), and Wakhi (Iranian, Pamir). Given that there are languages belonging to different families, we may expect differences in the phonetic realization of stop laryngeal contrasts and the articulation of retroflexes across languages. The acoustic, articulatory, and aerodynamic data were collected in Chitral and Gilgit, Pakistan in 2017. A total of 33 participants were recorded (Balti: 5; Burushaski: 4; Kalasha: 9; Khowar: 5; Shina: 6; and Wakhi: 4). The results of this study will help understand the diverse nature of stop consonants in the endangered languages of Northern Pakistan.

1:35

1pSCa3. Evidence for active cavity expansion through advanced lingual place of constriction in voiced geminates. Ushasi Banerjee (The English and Foreign Lang. Univ., Hyderabad, Telangana 500007, India, banerjeeushashi@gmail.com), Indranil Dutta, and Meghavarshini Krishnaswamy (Computational Linguist, The English and Foreign Lang. Univ., Hyderabad, Telangana, India)

Geminate voicing requires both active and passive cavity expansion in order to delay the equalization of subglottal and supraglottal pressure (Ohala 1983, Ham 2001). Studies report differences in stop closure duration (Lahiri & Hankamer 1988) and pre-geminate vowel duration as cues to the laryngeal contrast. Following Schroeder (1967), Mermelstein (1967), and Iskarous (2010), we present results from formant transitions in the VC:V context and the anti-symmetric Fourier component coefficient, $a_n = -2\ (F_n\text{-}f_n)/f_n$, to partially recover the area function of the vocal tract behind the geminate constriction. Voiced geminates exhibit significantly higher and positive coefficient values than their voiceless counterparts which implies that measured F_n values for voiced geminates are lower than their neutral tract formant values. This finding indicates that voiced geminates have a more anterior constriction compared to their voiceless counterparts. Without ruling out the role of passive cavity expansion through the cheek muscles, we conclude that in order to delay the pressure equalization the lingual constriction in voiced geminates is maintained at an anterior location compared to voiceless geminates in Bangla. Results from the slopes of first order F_2 locus-equations are also presented to measure the differences in coarticulatory resistance between the voiced and voiceless geminates.

1:50

1pSCa4. A contrastive place of articulation between palatals and velars: An ultrasound study of Malayalam palatal-velar nasals. Dominique A. Bouavichith, Savithry Namboodiripad (Linguist, Univ. of Michigan, 611 Tappan St, #455C, Ann Arbor, MI 48109, dbouavichith@gmail.com), and Marc Garellek (Linguist, Univ. of California, San Diego, La Jolla, CA)

Malayalam (Dravidian) has been described as having seven contrastive nasal places of articulation: labial, dental, alveolar, retroflex, palatal, velar, and a "seventh nasal," sometimes called palatal-velar /ŋj/ (Asher & Kumari 1997; Namboodiripad & Garellek 2017). Nasal inventories of this size are cross-linguistically rare, and the palatal-velar nasal has only been proposed in a handful of languages worldwide (e.g., Irish, Yanyuwa, and Awngi). Indeed, its status in Malayalam could be deemed peripheral; it is infrequent and occurs in limited phonological contexts. Namboodiripad & Garellek (2017) claim that, though several consonants undergo palatalization in Malayalam, this palatal-velar nasal is dynamically stable and not post-palatalized. Although it is distinct in format transitions from the non-palatalized velar nasal, the articulatory basis for the distinction between this palatal-velar nasal and a velar-glide cluster is not clear. The present study uses ultrasound imaging to determine if this seventh nasal is in fact dynamically stable. Ultrasound data were collected from one native speaker of Malayalam; the placement and timing of lingual contact with the palate were examined. Discussion will focus on how the palatal-velar differs in terms of place and timing of articulations from both palatal and velar nasals, as well as from other palatalized consonants.

MONDAY AFTERNOON, 7 MAY 2018 NICOLLET A, 2:15 P.M. TO 5:15 P.M.

Session 1pSCb

Speech Communication: South Asian Languages II (Poster Session)

Kelly Berkson, Cochair
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Christina M. Esposito, Cochair
Linguistics, Macalester College, 1600 Grand Ave, St. Paul, MN 55105

Indranil Dutta, Cochair
Department of Computational Linguistics, English and Foreign Languages University, Tarnaka, Osmania University Campus, Hyderabad 500605, India

All posters will be on display from 2:15 p.m. to 5:15 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 2:15 p.m. to 3:30 p.m. and authors of even-numbered papers will be at their posters from 3:30 p.m. to 5:15 p.m.

Contributed Papers

1pSCb1. Indic and Dravidian retroflexes: An ultrasound investigation. Amanda Bohnert, Samantha Myers, James Smith, and Kelly Berkson (Lingustics, Indiana Univ., 1021 E. Third St., Dept. of Linguist - Mem 322E, Bloomington, IN 47405, kberkson@indiana.edu)

The languages of South Asia somewhat famously contain what might be deemed canonical retroflexes. These are often described as involving an extreme articulation with sub-apical post-alveolar or palatal contact, and imaging studies of varying types (electropalatography, MRI, ultrasound, etc.) suggest that this is an accurate characterization for Dravidian retroflexes (Kochetov *et al.* 2014, Krull and Lindblom 1996, Narayanan *et al.* 1999, Narayanan *et al.* 2004, Scobbie *et al.* 2013). Indic retroflexes reportedly involve less extreme articulations (Ladefoged and Bhaskararao 1983), but imaging data for Indic languages remains surprisingly rare. This raises an interesting question: how do Indic and Dravidian retroflexes compare articulatorily? We explore this question by presenting ultrasound imaging of retroflexes in several Indic (Marathi, Hindi) and several Dravidian (Kannada, Telugu) languages. Results indicate that all languages contain at least some of the extreme articulations often associated with South Asian retroflexes, though the within-speaker variability noted in previous work (e.g., Arsenault 2017) is also on display.

2:15

1pSCb2. Articulatory and acoustic investigation of coronals in Hakha Chin. James Smith, Stefon Flego, and Kelly Berkson (Lingustics, Indiana Univ., 1021 E. Third St., Dept. of Linguist - Mem 322E, Bloomington, IN 47405, kberkson@indiana.edu)

Coronal obstruents are profoundly common typologically, occurring in most or all of the world's known languages. Very few languages, however, contrast coronal obstruents at the dental and alveolar places of articulation. Hakha Chin—also known as Lai—is a Tibeto-Burman language spoken in western Myanmar that is reported to do so. Very little phonetic research on Chin exists, but Maddieson and Van Bik (2004) provide acoustic and static articulatory data (palatography and linguography) suggesting that the contrast is between a lamino-dental series and an apico-alveolar series. We follow up on that research using dynamic, volumetric articulatory imaging in the form of 3D/4D ultrasound. Articulatory and acoustic data from two native speakers of Chin are presented in order to contribute to a more thorough understanding of this typologically uncommon contrast.

1pSCb3. Acoustic correlates of the four-way laryngeal contrast in Marathi stops. Olga Dmitrieva (Purdue Univ., 640 Oval Dr., Stanley Coulter 166, West Lafayette, IN 47907, odmitrie@purdue.edu) and Indranil Dutta (The English and Foreign Lang. Univ., Hyderabad, Telangana, India)

As many of its geographical and typological neighbors, Marathi demonstrates a four-way laryngeal contrast in stop consonants, which includes plain voiced and voiceless stops as well as aspirated voiceless stops and typologically rare aspirated (breathy) voiced stops. The present study investigates the acoustic correlates of the laryngeal contrast in word-initial velar stops in Marathi. Thirty three native speakers of Marathi were recorded in Mumbai, Maharashtra, India. Positive and negative Voice Onset Time (VOT), fundamental frequency at the onset of voicing (onset f0), and spectral intensity as a measure of breathiness during the following vowel were investigated as correlates of laryngeal distinctions. Preliminary results show that all four stop categories are well-distinguished via VOT and onset f0 measures. Initial spectral intensity measurements suggest that breathiness in Marathi stops is largely contained in the release and does not extend far into the following vowel, unlike what has been found in Hindi (Dutta, 2007). The results will be discussed with respect to the typology of voicing contrast across languages, specifically the relevance of VOT distinctions and the use of onset f0, and compared to relevant findings for typologically similar languages.

1pSCb4. VOT in Tibetan is conditioned by tone. Christopher A. Geissler (Linguist, Yale Univ., 370 Temple St, New Haven, CT 06511, christopher.geissler@yale.edu)

Several South Asian languages exhibit interactions among tone and consonant phonation, including Tamang (Mazoudon 2014) and Gurung (Ronkos 2015). In Central Tibetan, oral stops present a four-way contrast between two voicing/aspiration categories split across two lexical tones, high and low/rising. Across tones, previous analyses propose either a two-way contrast between unaspirated and aspirated stops (e.g., Dawson 1980), or a three-way contrast between voiced, unaspirated, and aspirated series (e.g., Denwood 1999), though acoustic measurements are lacking. This study recorded 19 native speakers of Tibetan in Kathmandu, Nepal, producing words exemplifying the phonological contrasts in stops and high/low tones. VOT measurements showed similar values for the unaspirated stops in both tonal environments, but a longer VOT in high-tone aspirated stops (46 ms) than low-tone aspirated stops (30 ms), or three different positive VOT values. That this three-way surface contrast cross-cuts tonal categories suggests an underlying two-way aspiration contrast. The apparent contrast between middle-VOT and long-VOT series results from the presence of a phonologically-specified high tone, which causes a lengthened VOT. This provides evidence for the conditioning of apparently-segmental characteristics by tone.

1pSCb5. Within-category variation in production of Hindi and English stop consonants. Ivy Hauser (Linguist, Univ. of Massachusetts Amherst, 650 North Pleasant St, Amherst, MA 01060, ihauser@linguist.umass.edu)

This paper addresses the broad question of how phonological structure affects phonetic production using within-category variation of multiple phonetic cues in Hindi and English stop consonants. The intuitive hypothesis, formalized in Dispersion Theory (Lindblom, 1986) is that languages that have more phonological contrasts along a single phonetic dimension should have relatively less within-category variation along that dimension. Hindi (four stop contrasts per place of articulation) speakers are therefore predicted to exhibit less within-category variation in production relative to English (two contrasts per place) speakers. This was observed in the results of Hauser (2016) who examined VOT variation of voiceless aspirated stops in these languages. The present study builds on those results and examines F0 as a second cue to the voicing contrast. The F0 results do not conform to the prediction that Hindi speakers should exhibit less variation. There was no difference in F0 variation between the Hindi and English speakers, yet there were significant differences in the amount of within-category F0 variation among the individual Hindi speakers (Levene's Test $p < 0.001$). It is argued that this is because F0 acts as a secondary cue to the voicing contrast. Speakers can exhibit greater and individualized levels of variability in F0 without adding perceptual confusion because the primary cue (VOT) remains distinct.

1pSCb6. Phonation type in vowels of heritage and native Gujarati speakers. Kiranpreet Nara (Linguist, Univ. of Toronto, 100 St. George St., Toronto, ON M5S 3G3, Canada, kiranpreet.nara@mail.utoronto.ca)

Gujarati, an Indo-Aryan language, contrasts phonation type in vowels: [baɾ] 'twelve' and [ba..ɾ] 'outside' (Pandit, 1957) and the current study looked at Gujarati vowels using acoustic and electroglottographic (EGG) analyses. The participants were native and heritage Gujarati speakers. Heritage speakers were born in Canada or arrived in Canada before seven years of age and learned Gujarati as their first language. Due to limited access to their first language, such as listening more than speaking the language and using Gujarati exclusively at home, it was expected that there might not be a significant difference between the heritage speakers' breathy and modal vowel productions. This study determined if the acoustic and EGG parameters that differentiate breathy from modal vowels were the same or different for both speaker groups. Some of the parameters used to distinguish phonation type were H1-H2, H1-A1 ('A1' refers to amplitude of the first formant), harmonic-to-noise ratio, and contact quotient. Measurements were made using VoiceSauce (Shue *et al.*, 2009) and EGGWorks (Tehrani, 2009). ANOVA analyses conducted on vowels /a e o/ indicated that fewer parameters distinguish phonation type for heritage than native speakers and the difference between breathy and modal vowels for heritage speakers is smaller in magnitude. The results thus far indicate that heritage speakers acquire a reduced phonation type contrast.

1pSCb7. Pre-coda vowel duration in Nepali. Martha Schwarz (Linguist, UC Berkeley, 6449 Colby St, Oakland, CA 94618, martha_schwarz@berkeley.edu)

Many previous studies have found that vowels are longer before voiced consonants than voiceless (Chen 1970). Fewer studies have examined vowel duration preceding aspirated vs. unaspirated stops, and results are inconsistent (Maddieson & Grandour 1976, Ohala & Ohala 1992). Durvasala and Luo (2012) find that both voicing and aspiration have a lengthening effect on the preceding vowel in Hindi, and that the voiced aspirated class has the longest preceding vowel duration. In the present study, I measure the duration of vowels preceding word final codas in Nepali, an Indo-Aryan language with the same four-way contrast as Hindi. A phonetic study of final stop realization has not yet been reported in the literature, though Nepali is described as variably neutralizing voiced aspirated to the plain voiced. I present an acoustic analysis of cues in word-final position including voicing, burst, and preceding vowel duration. I find evidence of neutralization of voiced and voiced aspirated stops in terms of aspiration duration, but also find results consistent with Durvasala and Luo (2012), that the voiced aspirated stops have longer preceding vowel than the voiced. The difference in vowel duration between the voiced and voiced aspirated classes suggests incomplete neutralization of voiced aspirated to voiced.

1pSCb8. Do Hindi speakers of English realize the acoustic cues for American English /v/ and /w/ contrast? Vikas Grover (Commun. Disord. and Deafness, Kean Univ., New Jersey, NJ 07083, vgrover@kean.edu), Valerie Shafer (Speech-Language-Hearing Sci., The Graduate Ctr., CUNY, New York, NY), D. H. Whalen (Speech-Language-Hearing Sci., CUNY Graduate Ctr., Haskins Labs. and Yale Univ., New York, NY), Erika Levy (Commun. Sci. and Disord., Teachers College, Columbia Univ., New York, NY), and Stephanie Kakadelis (Linguist Program, The Graduate Ctr., CUNY, NY, NY)

The Hindi phonological inventory does not include a [v-w] contrast. In a perception study, Hindi speakers identified /v/ and /w/ in American English (AE) words with approximately 50% accuracy (Grover *et al.*, 2016). The present study compares AE /v/ and /w/ productions in bilingual Hindi-English speakers' with those of AE speakers' in nonsense words. Two AE listeners annotated recordings of word-initial and word-medial tokens to label F2 onset and presence of frication. F2 onset frequency and F2 slope were calculated. Hindi speakers were divided into two groups, those with length of residence (LOR) in the US of more than five years, and those who reside in India. Results revealed a significant difference between the AE group and both Hindi groups for F2 onset frequency and F2 slope. There was no significant difference between Hindi groups. The AE group had frication for /v/ tokens, but not for /w/. Both Hindi groups produced frication at about the same rate for /v/ and /w/. The finding of no significant difference between the Hindi groups indicates that exposure to the /v/-/w/ contrast in the US did not improve Hindi speakers' production. This further indicates a need to design targeted training for this contrast.

1pSCb9. Consonant-f_0 interaction in Malayalam stops: Automatic or controlled? Indranil Dutta and Molly Varghese (Dept. of Computational Linguist, The English and Foreign Lang. Univ., Hyderabad, Telangana 500007, India, indranil@efluniversity.ac.in)

Voicing is known to perturb f_0 in the following vowel (House and Fairbanks, 1953). This lowering has been attributed to biomechanical constraints (Svantesson and House, 2006). On the basis of evidence from languages where the [voice] feature does not manifest itself phonetically, yet is accompanied by f_0 lowering, it has been argued that f_0 lowering serves to enhance the phonological [voice] contrast. We present results from an acoustic study of f_0 perturbation in Malayalam, which exhibits predictable voicing in lexical items from its Dravidian stratum. Underlying voiceless stops only appear initially, and are voiced intervocalically. Malayalam also contains a large Sanskrit stratum lexicon, where phonetic voicing is tied to its stratal affiliation. We obtained a significant effect of laryngeal setting on z-score normalized f_0 at vowel onset. These results suggest voicing is significant in determining the onset f_0 of the vowel, regardless of the phonological status of voicing in a language We conclude that f_0 lowering has physiological and biomechanical bases. In that sense, f_0 lowering is an automatic consequence of the biomechanics of voicing. At the representational level, such lowering could indeed be enlisted for enhancement, however, that is not the only, or the primary purpose of f_0 perturbation.

1pSCb10. Relation of musical experience to perceiving and producing aspirated Hindi consonants. Anmol Gupta and Terry L. Gottfried (Psych., Lawrence Univ., 711 E. Boldt Way, Appleton, WI 54911, gottfrit@lawrence.edu)

Past research shows that those with musical experience often perceive and produce unfamiliar speech contrasts better than non-musicians (e.g., Gottfried, 2007; Lee & Hung, 2008; Marie, Delogu, Lampis, Belardinelli, & Besson, 2011; Slevc & Miyake, 2006). The present research seeks to clarify the mechanisms which underlie this relationship through a "test-teach-test" paradigm (see Vlahou, Protopapas, & Seitz, 2012). In the first session, native speakers of American English with and without conservatory training were tested on their discrimination of Hindi sentences that contained either an aspirated (e.g., [b^h]) or unaspirated (e.g., [b]) consonant ("test"). In a second session, participants completed training on producing and perceptually discriminating the aspirated and unaspirated Hindi consonants ("teach"). In the third session, participants were tested on their ability to perceive and to produce these unfamiliar consonants in the same and different phonetic contexts ("test"). Preliminary results suggest that all participants benefited from the training, but Improvement was markedly greater for musicians than for non-musicians. Current studies are further testing additional musicians and non-musicians to evaluate the extent which musical experience influences perception and production of these unfamiliar Hindi phonemes.

1pSCb11. Effect of contrastive focus on vowel duration in Bangla. Bitapi Ghosh (Dept. of Linguist and Contemporary English, The English and Foreign Lang. Univ., Amrita Pritam Hostel, Hyderabad, Telangana 500007, India, ghoshbitapi92@gmail.com)

The present study looks at Bangla, spoken near Kolkata, West Bengal, which has got the status of standard Bangla in present era, belongs to the Indo-Aryan branch of the Indo-European language family. Vowel length distinction is not phonemic in Bangla. However, the impact of various prosodic positions and information structure is quite obvious on the phonetic duration of the vowel. This paper presents an acoustic study of the effect of Contrastive focus on the duration of the high vowels /i/ and /u/ in Bangla. Six monosyllabic and six disyllabic words with the target vowels, altogether 24 words have been recorded in both Contrastive focus and Neutral focus contexts. 24 sentences as response to yes-no questions and for Neutral focus, 24 simple declarative sentences were recorded from six speakers of Standard Kolkata Bangla between age group of 20-25. In Bangla, Focus is realized by low rising pitch pattern where the f_0 movement have low f_0 valley followed by a rise (Hayes and Lahiri 1991). The present study throws light on the other acoustic cue. The statistical analysis of the results from the study show significant difference in vowel duration of the focussed segment.

1pSCb12. Quantifying the manipulation of F0 as an acoustic correlate of stress in Rebkong Amdo Tibetan. Nancy J. Caplow (English, Oklahoma State Univ., 205 Morrill Hall, Oklahoma State University, Stillwater, OK 74078, nancy.caplow@okstate.edu)

Quantifying the manipulation of F0 as an acoustic correlate of stress in Rebkong Amdo Tibetan Rebkong Amdo is a non-tonal variety of Tibetan spoken in Rebkong County, Qinghai Province, PRC. Speakers draw on varied acoustic resources to convey stress in disyllabic words, which are highly frequent in the language. Non-verbs (nouns, adjectives, and numerals) are stressed on the second syllable (S2); the primary acoustic correlate of stress is F0. Verbs are stressed on the first syllable (S1); stress is conveyed by both intensity and F0. For non-verbs, speakers manipulate F0 in different ways to convey S2 prominence. The pitch contour may (a) be flat in both syllables but higher in S2; (b) be flat or slope gently in S1 but fall sharply in S2; or (c) be generally flat in S1 but define a prominent arcing curve in S2. These three patterns, which occur in free variation, require different analytical approaches in order to quantify relative acoustic prominence across syllables: (a) mean F0 (Hz) is measured for each syllable; (b) pitch slope (in Hz/100 ms) is measured for each syllable; (c) F0 curves are characterized using a principal components analysis, and by determining the tonal center of gravity. In each approach, measurements are paired to determine the magnitude of the F0 contrast across syllables in individual words; these are then aggregated, demonstrating that speakers' use of F0 to convey S2 prominence is statistically as well as perceptually significant.

1pSCb13. Glottal phonation in Punjabi: Tone or aspiration. Paroma Sanyal and Manujata Midha (Dept. of Humanities and Social Sci., Indian Inst. of Technol. Delhi, New Delhi, Delhi 110016, India, psanyal@iitd.ac.in)

This study investigates a puzzle in the glottal phonation of the South Asian Language Punjabi. Though phonetically absent from the spoken language, voiced aspirated plosives (VAP) are still perceived as a distinct phoneme by native speakers. This phonological identification is achieved by the introduction of rising and falling tones in complementary phonological environments. Specifically, the VAP lose both voicing and aspiration in word-initial positions and yet speakers are able to discern the expected phoneme due to the epenthesis of a falling tone. Elsewhere, the VAP is phonetically indistinct from the corresponding voiced plosives, but for the rising tone on the same syllable. The vowel bearing the inserted tone in Punjabi must also be stressed. Consequently, the underlying VAP and the corresponding tones can be heterosyllabic. However in such scenarios, the tones associated with the position of VAP is reversed. We propose that an attempted glottal friction is linked to the introduction of the rising and falling pitch contour. In turn, this tone is also correlated with stress, which phonetically corresponds to low pitch in Punjabi.

1pSCb14. Predictions of Laryngeal Realism for breathy-voiced stops: Counter-evidence from Bangla. Jahurul Islam (Linguist, Georgetown Univ., Poulton Hall, 1421 37th St. NW, Washington, DC 20057, mi302@georgetown.edu)

The Laryngeal Realism (LR) framework (Honeybone 2005, Beckman *et al.* 2013) argues that the breathy-voiced stops in the four-way stop systems are *specified* for multiple features: [voice] and [spread glottis] ([sg]). As a diagnostic of the *specified* features, Beckman *et al.* (2013) proposed that the acoustic cues for a specified feature be longer in slower speech. While predictions of the LR framework have been tested on two- and three-way stop systems using this technique, there has not been much on the four-way stop systems. This study presents evidence from Bangla and argues that the predictions of the LR framework apply variably for the breathy-voiced stops; [sg] is found to be the stronger candidate as the specified feature than [voice]. In slower speech, the cue for the breathy voicing (the lag-time) lengthened in the breathy-voiced stops. Contrarily, the acoustic cue for the [voice] feature (the prevoicing) was not affected by speech rate. Mixed-effects models confirmed that the effect of speech rate on the duration of breathy-voicing cues was statistically significant, but the effect on prevoicing was not. These results imply that the prediction of the LR framework may not be the best representation of the phonological features of breathy-voiced stops.

1pSCb15. The length of copied consonants in geminates, total assimilation, and ambisyllables. Mary Burke and Melissa A. Robinson (Linguist, Univ. of North Texas, 3940 N. Elm St, Ste. B201, Denton, TX 76207, burkemk412@gmail.com)

Lamkang is an under-documented endangered Kuki-Chin language of the Northeast India spoken in 40 villages spread across Manipur's Chandel district with under 10,000 Lamkang-Naga speakers (Ethnologue 2014). In Lamkang, doubled consonants have three different sources: (1) gemination when stems are followed by enclitics of the shape V; (2) total assimilation of enclitics to stem codas; (3) ambisyllables in prefixes of the shape CV (where V is an epenthetic vowel). We examine the acoustic properties of the doubled consonants in these three morphological environments to show that the length of the closures reflect the morphological derivational history. The paper investigates a method for measuring doubled consonant length: We measure from peak intensity of vowels on either side of the consonant closure, rather than the length of the closure. Orthographic practice reflects these differences as doubled consonants in the prefixes are shorter than the phonologically derived consonants with enclitics. Copied consonants are written for enclitics but not in prefixes.

MONDAY AFTERNOON, 7 MAY 2018 — GREENWAY H/I, 1:00 P.M. TO 5:15 P.M.

Session 1pSP

Signal Processing in Acoustics, Underwater Acoustics, and Architectural Acoustics: Signal Processing for Complex Environments II

Sandra L. Collier, Cochair
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Kainam T. Wong, Cochair
Dept. of Electronic & Information Engineering, Hong Kong Polytechnic University, 11 Yuk Choi Road, Hung Hom KLN, Hong Kong

Max Denis, Cochair
U.S. Army Research Lab., 2800 Powder Mill Road, Adelphi, MD 20783-1197

Invited Papers

1:00

1pSP1. Performance of WUW and general ASR speech recognition systems in different acoustic environments. Gamal Bohouta and Veton Këpuska (Elec. & Comput. Eng. Dept., Florida Inst. of Technol., 220 E University Blvd. Apt. 701, Melbourne, FL 32901, gidris2015@my.fit.edu)

The Automatic Speech Recognition (ARS) Systems have become popular in the different areas of applications, such as education assistance, medical assistance, personal assistant, robotics and vehicles, telecommunications Industry, disabilities systems, home automation, and security access control. All these applications in life will increase the number of speech commands, which are Wake-up-words. The WUW speech recognition is similar to Key-Word spotting. However, in this paper, we propose an approach that will be used to detect the Wake-up-word (WUW) as a command with 100% accuracy and to provide a whole speech recognition system that can work in both WUW and General ASR systems with high accuracy. Also, this new ASR system will be used to solve one of the biggest problems in speech recognition, which is how to discriminate between the uses of a word or phrase in an alerting and a referential context. For example, in the use of the word "Computer'' in an alerting context is "Computer, show me the chart? '' and in a referential context it is "Every computer should have a speaker. Moreover, we tested the new system with different acoustic environments, such as different background noise levels, different speaker distances to the microphone, and different speakers.

1:20

1pSP2. Automatic target recognition with uncertain scattered signal distributions. D. K. Wilson, Carl R. Hart (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd, Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil), Chris L. Pettit (U.S. Naval Acad., Annapolis, MD), Daniel J. Breton (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., Hanover, NH), Edward T. Nykaza (Construction Eng. Res. Lab., U.S. Army Engineer Res. and Development Ctr., Champaign, IL), and Vladimir E. Ostashev (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

One of the primary challenges for performing robust automated target recognition (ATR) is how to compensate the signatures for environmental propagation effects. ATR algorithms tend to function well only in the specific terrain and atmospheric conditions for which they were trained. This is particularly true for acoustic signals, which undergo strong frequency-dependent scattering and refraction in both outdoor and underwater environments. To address this problem, we formulate a Bayesian sequential updating method, which accounts for realistic signal and noise distributions, and uncertainties in the parameters of these distributions. The formulation utilizes physics-based scattering models for signal fading and the cross coherence between frequencies and transmission paths. We discuss how, in a Bayesian context, the scattering models correspond to likelihood functions, which are conveniently paired with their conjugate priors to efficiently update the uncertain signal parameters (hyperparameters). The original (prior) distributions, as predicted using an initial forecast based on available weather and terrain data, are nudged towards the observed signal properties as samples are collected in real time. This leads to an optimal posterior distribution, as appropriate for use in a Bayesian classifier. The technique is demonstrated using simulations in which signal transmissions are collected along single and multiple transmission paths.

1:40

1pSP3. Active acoustic intensity errors due to contaminating noise. Mylan R. Cook, Kent L. Gee, Tracianne B. Neilsen, and Scott D. Sommerfeldt (Phys. and Astronomy, Brigham Young Univ., N201 ESC, Provo, UT 84602, mylan.cook@gmail.com)

Bias errors for both one and two-dimensional intensity probes have been previously calculated for both the traditional cross-spectral and the Phase and Amplitude Gradient Estimator (PAGE) methods [E. B. Whiting *et al*., J. Acoust. Soc. Am.**142**, 2208–2218 (2017); E. B. Whiting, M. S. Thesis (2016)]. Here, these calculations are expanded to include errors due to the source angle relative to the probe and due to contaminating noise. The noise can either be uncorrelated at each microphone or self-correlated, which is modeled as a plane wave with a varying angle of incidence. The errors in both intensity magnitude and direction are dependent on the signal-to-noise ratio (SNR), frequency, source properties, incidence angles, probe configuration, and processing method. The PAGE method is generally found to give more accurate results, though uncorrelated noise with a low SNR can yield large errors. For broadband signals, the PAGE method can be used beyond the spatial Nyquist frequency. [Work supported by NSF.]

2:00

1pSP4. Noise adaptive real-time estimation of infrasound using a compact array. William G. Frazier (Hyperion Technol. Group, Inc., 3248 West Jackson St., Tupelo, MS 38801, gfrazier@hyperiontg.com)

Previous investigations [Frazier, ASA Pittsburg (2015); ASA San Francisco (2013); ASA Kansas City (2012)] have demonstrated that exploiting the correlated statistical properties of wind noise on short spatiotemporal scales with a compact array permits enhanced detection and reduced estimation error of infrasound transients resulting from explosions. The approach was shown to be effective in real-time using non-parametric wind noise models. More recently [Frazier, ASA Boston (2017); ASA Salt Lake City (2016)], the basic ideas of the approach were extended to estimation of quasi-stationary infrasound signals such as microbaroms using parametric wind noise models and was demonstrated to be effective, but the computational requirements were too demanding for real-time implementation. This presentation will highlight modifications that enable real-time implementation for quasi-stationary signal estimation.

2:20

1pSP5. A maximum likelihood estimator for spectral models of acoustic noise processes. Gregory W. Lyons (Construction Eng. Res. Lab., U.S. Army Engineer Res. and Development Ctr., Engineer Res. and Development Ctr., 2902 Newmark Dr., Champaign, IL 61822, gregory.w.lyons@erdc.dren.mil) and Carl R. Hart (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

The power spectral density of time series from many acoustic phenomena can vary rapidly through several decades of magnitude, particularly for noise processes. This property can complicate evaluation of spectral density models with respect to a non-parametric estimate of the spectral density, also known as the periodogram. Typical measures, such as the sum of squares of the residuals, can suffer from bias errors and oversensitivity to large values. A log-likelihood function is here presented for a periodogram derived from a sequence of independent, circularly-symmetric complex normal Fourier components. The bias-corrected Akaike's information criterion of an expected-value spectral model is shown to be a useful measure for multi-model selection. A maximum-likelihood estimator is formed for spectral model parameters with respect to a known periodogram, along with approximate confidence intervals for the parameter estimates. Results from the maximum-likelihood estimator are compared with weighted and unweighted least-squares methods by estimating model parameters for periodograms from both synthetic and measured noise time series. Practical considerations for general periodogram fitting and numerical methods are discussed.

2:40

1pSP6. Explorations of *in-situ* passive source localization in the Philippine Sea using frequency-difference matched field processing. David J. Geroski (Appl. Phys., Univ. of Michigan – Ann Arbor, Randall Lab., 450 Church St., Ann Arbor, MI 48109, geroskdj@umich.edu), Matthew Dzieciuch (SIO/UCSD, La Jolla, CA), and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Matched Field Processing (MFP) is a well-known technique for source localization in challenging acoustic environments. Accurate source localization using this method requires a correlation between array-recorded and replica acoustic fields calculated using knowledge of the acoustic environment. For a well-understood acoustic environment, this method is successful. However, recorded signals may be sensitive to details of the environment which are unknown and not included in the calculation of the replica field. The severity of this mismatch increases with frequency, as phase coherence is lost more easily at higher frequencies. In this case, a mismatch between the recorded and calculated fields will occur which can cause MFP to fail at relevant frequencies and ranges in the deep ocean. A proposed remedy to this problem may be analyzing the frequency-difference autoproduct of the measured acoustic field instead of analyzing the field itself. This presentation explores using this method to localize moored sources using data from the North Pacific Acoustic Library. Localization statistics are presented based on sources which are moored hundreds of kilometers from the receiving array, broadcasting between 200 and 350 Hz, and are matched using out-of-band frequencies between 10 and 50 Hz. [Sponsored by ONR.]

3:00–3:15 Break

3:15

1pSP7. Signal and information processing in complex ocean environments. David P. Knobles (KSA, LLC, PO Box 27200, Austin, TX 78755, dpknobles@yahoo.com), Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Mohsen Badiey (Elec. Eng., Univ. of Delaware, Newark, DE)

Even a cursory examination of the information theory relationship between model capacity/complexity, model error, and prediction error should suffice to convince one that to treat the ocean as deterministic by increasing the dimensionality of the number of physical mechanisms, parameters, etc., may prove ineffective in achieving desired results in complex waveguides. Instead of a deterministic approach that results from solving a high-dimensional inverse problem which can lead to poor predictive capability, the approach advocated here predicts the statistics of the Detection, Classification, and Localization (DCL) problem. Various dimensional reduction methods are utilized such as those that can be applied to images in the spectral acoustical domain. Bayesian Information Criteria (BIC) and trans-dimensional methods coupled with machine learning algorithms can then be used to construct a meaningful signal processing approach which optimizes predictive capability. Acoustic and environmental survey data collected in the New England Seabed Characterization experimental area that has a high horizontal variability are used to demonstrate the approach. [Work supported by ONR.]

3:35

1pSP8. Temporal-spatial and source level distributions for a repertoire of fin whale vocalizations in the Norwegian Sea observed with a coherent hydrophone array. Heriberto A. Garcia (Elec. and Comp Eng., Northeastern Univ., 151 Mystic St., Apt 35, Arlington, MA 02474, garcia.he@husky.neu.edu), Delin Wang, Chenyang Zhu (Elec. and Comp Eng., Northeastern Univ., Boston, MA), WEI HUANG (Elec. and Comp Eng., Northeastern Univ., Malden, MA), Anna Kaplan, and Purnima Ratilal (Elec. and Comp Eng., Northeastern Univ., Boston, MA)

The vocalizations of the fin whale are detected, characterized and localized over instantaneous wide areas of the Norwegian and Barents Seas using a large-aperture densely-sampled coherent hydrohone array via the passive ocean acoustic waveguide remote sensing (POAWRS) technique from observations in late winter to early spring 2014. The fin whale vocalizations are comprised of their characteristic 20 Hz pulses, interspersed by 130 Hz upsweeps, less abundant 30-100 Hz downsweep chirps and 18-19 Hz centered backbeats. The time-frequency characteristics of these vocalization types and their diel occurrence rate time-series are quantified in three distinct regions of the Norwegian Sea, off the coasts of Alesund, Lofoten, and the Northern Finnmark. Their vocalization rates are a factor of roughly 5 times and 17 times larger respectively for the 20 Hz and 130 Hz centered vocalizations off Northern Finnmark than the other regions. The vocalization rate spatial density distributions, source level distributions and probability of detection (PoD) regions are estimated for the 20 Hz and 130 Hz vocalizations. Highly intense 20 Hz pulses from closeby fin whales are received with their first-order harmonics centered at 40 Hz with twice the bandwidth but 30 dB lower source levels, indicating a nonlinear mechanism for their production.

3:55

1pSP9. Design of a signal normalizer for high-clutter active-sonar detection. Frank W. Bentrem, Jonathan Botts (Appl. Res. in Acoust., 209 N. Commerce St, Ste 300, Culpeper, VA 22701-2780, frank.w.bentrem@ariacoustics.com), and Jason E. Summers (Appl. Res. in Acoust., Washington, District of Columbia)

We present normalizer design considerations for detection of targets in a high-clutter environment. Several classes of constant-false-alarm-rate (CFAR) normalizers have conventionally been used to bring out a target signal from the background noise and reverberation in active sonar signal processing. The special challenges of detection in high-clutter environments are outlined, and the normalizer classes are compared for their suitability of use in these environments. We consider statistical and multiple-look normalizers and give emphasis to the class of split-window mean-power normalizers and implementations that preserve target signal-to-noise ratio (SNR) and highlight structure while mitigating artifacts that are sometimes produced near clutter edges. Statistical comparisons and specific examples are presented, and we propose an efficient CFAR normalizer for high performance in regions of high clutter.

4:15

1pSP10. Dimension-reduced adaptive pulse compression with model error compensation. Jonathan Botts (Appl. Res. in Acoust., 209 N. Commerce St, Ste 300, Culpeper, VA 22701, jonathan.botts@ariacoustics.com), Jason E. Summers (Appl. Res. in Acoust., Washington, District of Columbia), Frank W. Bentrem (Appl. Res. in Acoust., Culpeper, VA), and Charles F. Gaumond (Appl. Res. in Acoust., Washington, District of Columbia)

In active acoustic imaging, range sidelobes from competing clutter and interferers in acoustically complex environments may mask signals of interest. In order to unmask these signals, increase range resolution, and decorrelate the time-series we introduce dimension-reduced adaptive pulse compression (APC) with model-error compensation that operates after a conventional matched filter. Structured covariance estimates minimize the required sample support while dimension reduction occurs naturally and a priori after the matched filter. We compensate for predictable signal distortions such as Doppler through both covariance matrix tapers and truncation of the covariance matrix. Model-based APC is computationally intensive, scaling with the third or fourth power with replica length, so algorithmic modifications are introduced to reduce computation by orders of magnitude. We evaluate the algorithm's robustness and computational feasibility on simulated and archived data. [Portions of this material are based upon work supported by the Naval Sea Systems Command.]

4:35

1pSP11. Signal compression technique in decomposition of arrivals having different speed of sound propagation through human lungs in the frequency range of 10–19 kHz. Vladimir Korenbaum, Anton Shiryaev, Anatoly Kostiv, and Maria Safronova (Pacific Oceanologic Inst., 43, Baltiiskaya Str., Vladivostok 690041, Russian Federation, v-kor@poi.dvo.ru)

The unexpected phenomenon of sound transmission through human lungs at frequencies above 10 kHz with a speed of about 1000 m/s was revealed by Rueter *et al.*, 2010. The objective is a study of characteristics of sound transmission in human lungs in the frequency range of 10–19 kHz using signal compression technique. The 14-channel receiving apparatus was used. Chirp signals 10–19 kHz (6 min) were emitted into human thorax by small shaker. Sound propagation in human lungs was studied in paths with opposite chest positions of shaker and sensors in 4 volunteers. An existence of low-speed arrivals with propagation velocities of 150–50 m/s, which amplitude and/or velocity is inversely dependent on the air-filling of lungs (inspiration/exhalation) has been revealed. These arrivals may be treated as a result of sound propagation mainly through the lung parenchyma. On the contrary, the amplitudes of high-speed arrivals with velocities of 150–1000 m/s are enhanced with a decrease in air-filling of lungs. They may be connected to the sound propagation mainly through high-density tissues of thorax. The results are promising for medical acoustic visualization of local reduction in air-filling/ventilation of lung parenchyma. [This study was supported by the RFBR grant 16-08-00075.]

4:55

1pSP12. Processing of emitted respiratory noises to monitor health condition and spatial displacement of a scuba diver under water. Vladimir Korenbaum (Pacific Oceanologic Inst., 43, Baltiiskaya Str., Vladivostok 690041, Russian Federation, v-kor@poi.dvo.ru), Sergey Gorovoy (Far Eastern Federal Univ., Vladivostok, Russian Federation), Anatoly Kostiv, Anton Shiryaev, Aleksey Borodin, Veniamin Dorozhko (Pacific Oceanologic Inst., Vladivostok, Russian Federation), and Andrey Fershalov (Far Eastern Federal Univ., Vladivostok, Russian Federation)

Monitoring health condition of a scuba diver and his displacement within the exploited water area is necessary to ensure the safety of diving. Respiratory noises, emitted by the diver into water, may be used for this purpose. In the hydroacoustic basin, powerful signals, having repetition frequency of about 0.12 Hz are found, which correspond to diver's respiratory rate. The main broadband signal of exhalation is concentrated in the frequency band of 150–1150 Hz. The high-frequency signal in the frequency band of 3.5–4.7 kHz is associated with inspiration. Thus, acoustic estimating the respiratory rate and the ratio of the inspiration/expiration durations are possible, which are important physiological parameters to assess a health condition of the diver. When registering in sea respiratory noises of scuba diver against the background noise, it is possible to trace the acoustic signs of respiratory maneuvers associated with the noise of floating bubbles in the spectrograms at distances up to 100–200 m. The same acoustic signs provide monitoring the displacement of a scuba diver by determining the delays of the maxima of the cross-correlation function at 2 hydrophones. Trails of the delays in correlograms are traced at distances up to 300 m.

Session 1pUW

Underwater Acoustics: Instrumentation for Underwater Acoustics

Aaron Darnton, Chair
Naval Undersea Warfare Center Division Keyport, 610 Dowel St, Keyport, WA 98345

Contributed Papers

1:45

1pUW1. Design considerations for a compact correlation velocity log. Thomas Blanford (Graduate Program in Acoust., The Penn State Univ., 1623 S. Ashwicken Ct, State College, PA 16801, teb217@psu.edu), Daniel Brown, and Richard Meyer (The Appl. Res. Lab., The Penn State Univ., State College, PA)

A Correlation Velocity Log (CVL) has some advantages (lower source level and operating frequency) over a Doppler Velocity Log (DVL) as a navigational aid for an unmanned underwater vehicle (UUV). A CVL provides a bottom referenced velocity estimate by estimating displacement using the incoherently scattered field from an acoustic projector and an array of hydrophones. A small low cost UUV generally operates in shallow water and has limited space and power available for a navigational aid, creating added constraints for the design of a CVL. The important design considerations (such as size, array geometry, operating frequency, and bandwidth) will be discussed as they relate to accuracy of the velocity estimate and operating range. [The authors acknowledge the financial support for this work by Lockheed Martin Undersea Systems.]

2:00

1pUW2. Underwater fiber optic transducer. Aaron Darnton (Naval Undersea Warfare Ctr. Div. Keyport, 610 Dowel St, Keyport, WA 98345, aaron.darnton@navy.mil)

Fiber optic sensors have been used in various configurations including interferometric, Bragg grating, and Rayleigh scattering approaches. In this work, the Rayleigh approach is considered for use as a hydrophone. This approach transmits light along a single fiber and utilizes the backscattered light to infer small strains in the fiber. Typically in-water use of this type of optical acoustic receiver has been limited due to the gross motion of the fiber in the water. The gross motion tends to wash out the acoustically driven strains of interest. Additionally, sensitivity can be relatively low in this approach. This work considers sensing on a section of fiber wrapped around a mandrel for the purpose of increasing sensitivity as well as reducing the effects of gross motion.

2:15

1pUW3. Experiments with the Underwater Bubble Low Frequency Sound Resonator. Andrey K. Morozov and Douglas C. Webb (Teledyne Marine Systems, 49 Edgerton Dr., North Falmouth, MA 02556, amorozov@teledyne.com)

An infra-sound source with a resonator in the form of an underwater bubble made from an elastic material is a simple engineering solution. The elastic membrane supports high volume displacement and a large radiation aperture. The Rayleigh-Plesset equation defines the dynamics of a spherical bubble including surface tension and viscous effects. The shape of a large bubble deforms due to gravitational effects. Its internal pressure oscillations are comparable with the difference of gravity pressure and these effects are part of its dynamics. The real Q-factor of a practical bubble is smaller than theoretical. The underwater bubble resonator was studied experimentally in the pool and at the Woods Hole Oceanographic Institution's dock. The research includes the 3D finite-element analysis and its comparison with the experimental data. The study gave the real values for the Q-factor, resonance frequencies and sound pressure levels. The experimental resonator has good performance with a maximum SPL 190 dB at the resonance frequency 6-12 Hz. It was shown that a cylindrical resonator can be towed with a speed of 8 knots. The research proved that this is a practical approach for a coherent low frequency sound source, which can find applications in a marine seismic survey.

2:30

1pUW4. Videos of ultrasonic wave propagation through transparent acrylic objects in water for introductory physics courses produced using refracto-vibrometry. Matthew R. Mehrkens, Benjamin A. Rorem, and Thomas M. Huber (Phys., Gustavus Adolphus College, 800 W College Ave, Saint Peter, MN 56082, huber@gac.edu)

This project used refracto-vibrometry to produce full-field videos of ultrasonic wave propagation in water; these videos could be incorporated into high school and college physics courses to illustrate wave behaviors. Refracto-vibrometry (RV) is an interferometric method for optically measuring sound waves. The measurement beam from a Polytec PSV-400 scanning laser Doppler vibrometer was directed through a water tank towards a stationary retroreflective surface. Acoustic wave fronts (density variations) which pass through the measurement laser cause variations in the integrated optical path length. By superimposing tens of thousands of scan points, videos enable visualization of the ultrasound propagation. In one experiment, ultrasonic waves in water (1484 m/s) are incident on a submerged acrylic block at a non-zero angle. The waves are refracted as they travel at about 2700 m/s through the block. Students can determine this speed of sound by measuring the angle of refraction and lateral deflection of the wave fronts in the video. In another experiment, ultrasonic waves passed through an acrylic cylinder, producing a Mach cone in water. Measurement of the Mach cone angle enables determination of the acrylic speed of sound. Links for videos and sample worksheets will be provided.

Payment of an additional fee is required to attend this session.

MONDAY AFTERNOON, 7 MAY 2018 NICOLLET D2, 7:00 P.M. TO 9:00 P.M.

Session 1eID

Interdisciplinary: Tutorial Lecture on Hearing Loss and the Future of Auditory Implants

Christina J. Naify, Chair
Jet Propulsion Lab, 4800 Oak Grove Dr, MS 157-316, Pasadena, CA 91101

Chair's Introduction—7:00

Invited Paper

7:05

1eID1. Hearing loss and the future of auditory implants. Andrew J. Oxenham (Univ. of Minnesota, 75 E River Parkway, Minneapolis, MN 55455, oxenham@umn.edu)

Hearing loss is a major and growing health concern worldwide. According to the National Institute on Deafness and Communication Disorders (NIDCD), 17% of the adult population in the US (around 36 million people) report some form of hearing loss, with the proportion of affected individuals rising to nearly 50% among those aged 75 or older. Loss of hearing has been associated with increased social isolation, more rapid cognitive decline, and other more general health issues, although causal relationships have yet to be established. This tutorial will review the physiology of hearing loss, along with its perceptual consequences, as measured in the laboratory and experienced in everyday life. The focus of the tutorial will be on implantable technologies that have been used to alleviate severe hearing loss and deafness, with particular emphasis on cochlear implants. Although cochlear implants have enjoyed remarkable success over the past few decades, they do not restore normal hearing, and may be approaching their technological limits in terms of the benefits that patients can gain from them. The tutorial will end by exploring future directions of implantable and other technologies in the quest to restore and maintain hearing throughout the lifespan.

MONDAY EVENING, 7 MAY 2018 SKYWAY B, 5:00 P.M. TO 6:00 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, 8 May 2018.

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 ASA Committee on Standards: Acoustics of Lobbies, Atria, Stairways, Corridors, Pre-function, etc.

Logan D. Pippitt, Chair
Marvin Hall, 1465 Jayhawk Blvd., Lawrence, KS 66045

Chair's Introduction—9:00

Invited Papers

9:05

2aAAa1. The face of the facility. Brandon Cudequest and K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd. Ste. 325, Westlake Village, CA 91362, bcudequest@gmail.com)

Lobbies and atria are a facility's initial destination, the point of departure, the information center, and the security checkpoint. They define a building's aesthetic, connect to the rest of the facility, require high speech intelligibility at key locations, and demand durability. The requirements can fluidly fluctuate throughout design, and the acoustics need to keep pace. This paper will discuss different projects where the lobby was central to a successful design. Highlights include a hospital atrium, wherein a balance was struck between speech intelligibility and speech privacy; an airport concourse where the prospect of tile grout lines was carefully considered; and a giant courthouse lobby with design goals at odds with design standards, and with bridges providing acoustical "shade" for security guards.

9:30

2aAAa2. Acoustically neglected and ignored building spaces. Megan Stonestreet (Architecture, Univ. of Kansas, University of Kansas, Lawrence, KS 66045, meganstonestreet@ku.edu)

Often neglected and ignored building spaces in regard to room acoustics and building noise are corridors, stairways, and lobbies. For example, reverberant corridors in school teaching buildings are often noisy and provide an undesirable acoustic connection between classrooms; stairways are typically reverberant spaces that act as vertical corridors and are generally unpleasant spaces; lobbies are actually multipurpose spaces that serve as entrance, exit, and gathering spaces, and for performance facilities, they are also used as performance spaces. Examples of each of these building spaces will be discussed along with acoustic data and practical acoustic requirements.

9:55

2aAAa3. Case studies of two atria to begin a conversation regarding the importance of acoustical performance of transitional spaces. Michelle Huey and Michael (Mick) Barnhardt (D.L. Adams Assoc., 1536 Ogden St., Denver, CO 80218, mhuey@dlaa.com)

Transitional spaces, such as lobbies or atria, are largely forgotten spaces regarding room acoustics. These transitional spaces are widely used for presentations, meetings, and parties/gatherings. Lobbies and atria are often connected to hallways and adjacent office clusters, where sound can propagate to areas far away from the source. With the ever-increasing need for transitional spaces to also act as gathering spaces, it is important to begin a discussion of acoustical standards in transitional spaces. This paper discusses two case studies conducted by D.L. Adams Associates: (1) An atrium with a kitchen and dining area used for parties, gatherings, meetings, and throughout the day by employees; (2) an atrium with a commercial-grade kitchen at the center used for meetings, lunchtime, gatherings, and special events. This paper includes discussions of noise propagation, noise build up, reverberation time, and absorptive treatments within transitional spaces and how to make the spaces work acoustically, while isolating them from adjacent areas.

10:20

2aAAa4. Case study of acoustics in two very large atriums. Timothy Foulkes (Cavanaugh Tocci Assoc., 327 Boston Post Rd., Sudbury, MA 01776, tfoulkes@cavtocci.com)

Two recent projects have involved very large spaces with predominantly hard finishes. One assignment was to develop an acoustic treatment plan for an existing six story interior courtyard within an office tower. This space, a very formal design finished in marble, wood paneling, and glass, had a 7 second reverberation time. The courtyard was available as a lounge space to all tenants of the building but was barely utilized. We were part of a design team hired by the building owners to update the courtyard and make it more user friendly, including improvement in the acoustics. The other atrium was part of a new project—a seven story glass enclosure with a concrete floor. This atrium serves many different functions including the main entrance and security lobby, dining space, informal meeting space, and lounge. The acoustics were mitigated by the irregular shape, areas of carpet on the balconies, and limited areas of acoustic

paneling. The new atrium is a success and it has become the living room and hub for this large building. It is well used throughout the work day for all of the purposes mentioned above and more. Renovations for the other space are still in design as of the abstract deadline, but there may be finished in time for the Spring ASA meeting. The presentation will include a discussion of the dimensions and finishes, acoustic priorities, design challenges, acoustic measurements, and subjective comments.

10:45

2aAAa5. Measured acoustic conditions in the common spaces of two educational buildings at the University of Nebraska–Omaha. Samuel H. Underwood and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St. Omaha, NE, Omaha, NE 68182-0816, samuelunderwood@unomaha.edu)

This paper presents results from an acoustical case study of the atria, corridors, stairways, and other common spaces within two buildings located on the campus of the University of Nebraska-Omaha: the Peter Kiewit Institute and the Barbara Weitz Community Engagement Center. Both buildings, while educational in nature, differ drastically in acoustical design. Impulse responses, noise levels, and transmission loss in both facilities are measured and the resulting acoustic metrics are compared to recommended values for the respective spaces. Descriptions of each building's architectural features are presented and linked to the measurement results. In particular, many spaces in the Peter Kiewit Institute were designed to expose the building systems, and the lack of acoustical products such as acoustical ceiling tile results in higher noise levels than normally expected. The Weitz Center, however, was designed with great emphasis placed on achieving optimal indoor environmental conditions. The results of this case study support the importance of appropriate acoustic design atria, corridors, stairways, and other common spaces in educational settings.

11:10

2aAAa6. The architectural rejuvenation of Canada's National Arts Centre: The function of pre-function. Robin S. Glosemeyer Petrone and Marcus R. Mayell (Threshold Acoust.com, 53 W Jackson Blvd., Ste. 815, Chicago, IL 60604, robin@thresholdacoustics.com)

When Canada's National Arts Centre opened in 1967, the Brutalist architecture style of its era yielded a fortress for the arts. As Canada celebrates its sesquicentennial, a $110M (Canadian) architectural rejuvenation process is transforming the inward facing fortress into alluring beacon at the heart of the capital. Expanded public spaces enveloped in transparent facades form multiple pre-function spaces that visually spill into one another to convey the constant activity of Canada's incubator for the performing arts to the patrons within and the surrounding city. The desire for visual transparency, the building's designation as a National Historic Site of Canada, and need for simultaneous programming of the performance and pre-functions spaces provided no shortage of acoustic challenges.

Contributed Paper

11:35

2aAAa7. Creating realistic design goals through the use of locally-measured reverberation time and background noise data. Matt Whitney and Ted Pyper (K2, 5777 Central Ave., Ste. 225, Boulder, CO 80301, matt@k2avt.com)

One of the major difficulties in the design process of an atrium or lobby is providing points of reference to both the architect and the end user to help drive the conversation on an acceptable reverberation time and background noise level for the projected use cases. K2 recently surveyed eight different atrium and lobby spaces in Denver and used this reverberation time and background noise data to develop an appropriate range of reverberation times, for a given room volume, covering most of the common uses for atriums and lobbies. These ranges help provide guidance to both the design team and the end user on design goals and expectations for these spaces. This presentation will present these data, as well as subjective impressions and architectural design of each of the spaces, and describe how K2 used this information to initiate productive conversations with architects and end users and produce realistic design goals.

Session 2aAAb

Architectural Acoustics: Student Design Competition (Poster Session)

David S. Woolworth, Cochair
Roland, Woolworth & Associates, 356 CR 102, Oxford, MS 38655

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 and the National Council of Acoustical Consultants are sponsoring the 2015 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 US$1250 will be made to the submitter(s) of the design judged "first honors." Four awards of US$700 each will be made to the submitters of four entries judged "commendation."

TUESDAY MORNING, 8 MAY 2018 LAKESHORE B, 8:00 A.M. TO 11:55 A.M.

Session 2aAB

Animal Bioacoustics and ASA Committee on Standards: History of Animal Bioacoustics

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

Chair's Introduction—8:00

Invited Papers

8:05

2aAB1. A Brief history of avian bioacoustics. Robert Dooling (Psych., Univ. of Maryland, Baltimore Ave., College Park, MD 20742, rdooling@umd.edu) and Micheal L. Dent (Psych., Univ. of Buffalo, Buffalo, NY)

A proper history of avian bioacoustics would reference early naturalists such as Aristotle, Pliny the Elder, and others who observed that birds not only have acute senses but that they also learn their vocalizations with reference to auditory information and exhibit parallels with human speech, language, and music. On the production side, the development of the tape recorder followed by the sound spectrograph enabled researchers to precisely record, preserve, analyze, and quantify the characteristics of the amplitude and spectral envelope of the acoustic signals which birds use to communicate. In the 1950s Thorpe, and later Marler, tackled the development of vocal learning in birds and the nuances of individual and species differences, which began the modern era of avian bioacoustics. Almost simultaneously, on the perception side, the confluence of Skinnerian conditioning methods, signal detection theory, and the first laboratory minicomputers launched the modern era of animal psychophysics, which continues to this day in helping to understand the striking parallels between avian acoustic communication and speech and language learning in humans.

8:25

2aAB2. A history of fish bioacoustics. Arthur N. Popper (Univ. of Maryland, Biology/Psych. Bldg., College Park, MD 20742, apopper@umd.edu) and Anthony D. Hawkins (Loughine Ltd, Aberdeen, United Kingdom)

Awareness of fish sound production dates back to ancient times, and concern about effects of man-made sounds on fishes can be traced back at least to the mid 17th century. By the end of the 19th century, the morphology of the fish ear had been well described, but experimental studies of the hearing characteristics did not begin until the early 1900s, when Parker demonstrated that fish can detect

sounds. Subsequent work by von Frisch and his students, including Dijkgraaf, determined the range of frequencies that fishes could detect, and showed the fishes can discriminate between sounds. Later work in Europe provided a deeper understanding of sound detection mechanisms with Enger pioneering experiments on fishes in the sea. Experiments in the sea by Schuijf, Chapman, and others demonstrated that fish could discriminate sound directions. Meanwhile, in the U.S., Moulton, Tavolga, and their students documented diversity in fish hearing capabilities. Concurrently, studies by Fish, Winn, Myrberg, Tavolga, and others showed diversity in fish sounds, and documented the behaviors associated with fish sound production. Work by these investigators (and many others) opened the field of fish bioacoustics, and provided a wealth of information that stands today as the basis for modern studies.

8:45

2aAB3. Amphibian bioacoustics: From Arch to Zelick. Peter M. Narins (Integrative Biology & Physiol., UCLA, 621 Charles E. Young Dr. S., Los Angeles, CA 90095-1606, pnarins@ucla.edu)

Amphibian bioacoustics may be thought to have originated with the first published acoustic playback experiments with frogs 60 years ago. Martof and Thompson (1958, Behavior 13:243–258) and Littlejohn and Michaud (1959, Tex. Jour. Sci.11:86–92) showed that male calls of *Pseudacris nigrita* in Georgia are effective in attracting conspecific females, and that females of *Pseudacris streckeri* in Texas can discriminate conspecific calls from heterospecific calls, respectively. Capranica (1965, JASA 40:1131–1139) was the first to electronically synthesize bullfrog calls and to electronically modify (filter) them for use as stimuli in acoustic playback experiments. Since then, all hell broke loose. Reel-to-reel, cassette and DAT tape recorders, were replaced by wideband portable digital machines capable of recording directly to CF cards and/or to high capacity internal hard drives. Sophisticated call synthesis and analysis programs are widely available. Amphibian communication has been studied on six continents and Madagascar and we have gained a more sophisticated appreciation of the production and reception of amphibian airborne signals. Now the challenge is to find new and interesting amphibian bioacoustic questions, and for inspiration, I shall present a brief review of some of the noteworthy experiments of the past. [Work supported by NSF grant #1555734.]

9:05

2aAB4. History of technology for studying animal sound and communication. David K. Mellinger (Coop. Inst. for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, David.Mellinger@oregonstate.edu)

New technology has always been used to listen to, study, and understand animal sounds. Early recording devices in the 1870s employing wax cylinders and discs were quickly employed for recording nature sounds. Underwater listening devices from the 1910s captured marine animal sounds; interest in these sounds increased greatly after World War II, when naval advances made underwater sound accessible to sonar operators who noted the presence of cetacean sounds. Microphone technology advanced to the point where ultrasonic and eventually infrasonic signals were recorded. Frequency separation and analysis was facilitated by the invention of the vocoder in the late 1930s, which separated sound into a number of frequency bands and enabled, for instance, the discovery of multiple sources of sound (more than one syrinx) in bird vocalizations. The greatest leap in frequency separation came with the discovery of the Fast Fourier Transform and its application on digital computers for making spectrograms; since the 1970s, these simple graphic displays have made it possible to look at sound signals statically, greatly facilitating analysis. Advances in digital multi-channel recording have improved our ability to localize animals and study communication networks. Advances in analysis software continues to improve our ability to interpret sounds of animals.

Contributed Paper

9:25

2aAB5. A brief history of our understandings on underwater noise impacts to marine life and the evolution of its regulatory process in the United States. Shane Guan, Amy R. Scholik-Schlomer, and Jacqueline Pearson-Meyer (Office of Protected Resources, National Marine Fisheries Service, 1315 East-West Hwy., SSMC-3, Ste. 13700, Silver Spring, MD 20910, shane.guan@noaa.gov)

It has been long understood that elevated noise is detrimental to signal detection by the anti-submarine warfare community. However, it was not until the early 1970s that similar concerns were raised regarding marine life that utilize sound for various life functions. The 1980s saw some of the first studies on effects of noise from offshore oil and gas exploration and development on marine mammals in Arctic waters. Low-frequency sources for ocean thermometry research and submarine detection over ocean basins brought additional concerns in the early 1990s on potential impacts on marine mammals from intense sources. Around late 1990s, several ad-hoc noise levels were adopted as acoustic impact thresholds of marine mammals by regulatory agencies in the U.S. Several cetacean mass stranding events that were coincidental with the mid-frequency military sonar operations and the increased awareness of ocean noise pollution accelerated research in this field in the 21st Century. Numerous studies on hearing sensitivity and noise induced temporary threshold shift or physical injuries on various marine animal taxa led to new sophisticated regulatory guidelines on assessing underwater noise impacts to marine life. In addition, recent understanding of soundscapes as environmental quality factors provides new perspectives on marine species and ecosystem conservation.

9:40–9:55 Break

9:55

2aAB6. The Office of Naval Research Marine Mammal Program, 1990–2006. Robert Gisiner (IAGC, 5410 Sycamore Villas Dr., Kingwood, TX 77345, gisiner@aol.com)

ONR first stood up a dedicated marine mammal research program in 1990. The initial effort was led by Dan Costa, with the support of experienced ONR program leaders like Mel Briscoe, Bernie Zahuranec, and Steve Zornetzer. The author was privileged to manage the program from 1993 to 2006; a period of rapid expansion of concern about the marine environment accompanied by the desire to put tools for improved understanding in the hands of scientists and decision makers. The emphasis from the start was on multi-disciplinary collaborations. Major research themes included hearing and exploration of Temporary Threshold Shift as a metric of auditory risk; advancement of animal-borne scientific instruments ("tags"); advancement of ocean acoustic monitoring technologies, refinement of field sound exposure experimental methods, and improvement of sound exposure models. Frequent external reviews, including four National Research Council (NRC) reviews, were vital to the quality and credibility of the program. The 2005 NRC review on the Population Consequences of Acoustic Disturbance (PCAD) is a textbook example of how ONR's basic research contributions have had far-reaching consequences that greatly exceeded initial expectations. The convergence of growing concern about manmade sound in the ocean and the ONR model for quality in basic research has had a profound impact on our ability to understand ocean bioacoustics and to make better informed decisions about the potential risks posed by human activities in the marine environment.

10:15

2aAB7. The last decade of bioacoustics at the Office of Naval Research. Michael Weise (Office of Naval Res., 875 N Randolph Sr, Arlington, VA 94598, michael.j.weise@navy.mil)

The goal of the ONR Marine Mammals and Biology (MMB) program is to enable Navy to and meet operational training and testing objectives in an environmentally responsible and legal manner. The ONR Marine Mammals and Biology program invests in Monitoring and Detection topic with the goal is to improve marine mammal monitoring capabilities over current methods. Develop new and existing technology such as passive acoustics, IR, and others. Recent research efforts on passive acoustics include the development and testing of new autonomous hardware platforms and signal processing algorithms for detection, classification, and localization of marine mammals. Ultimately, the ONR goal is to adapt those algorithms for use on a variety of fixed, towed, floating, and profiling platforms. For example, over the last several years ONR has adapted the use of autonomous ocean gliders for marine mammal monitoring to create the desired capability of persistent, autonomous, passive acoustic monitoring of an area for marine mammal presence and abundance to provide timely, reliable, accurate, and actionable information to support marine mammal mitigation and monitoring. A key goal of ONR sponsored technology development is making the technology available to the broader Navy and research communities.

10:35

2aAB8. The history of marine mammal bioacoustics at the Navy Marine Mammal Program. Dorian S. Houser (National Marine Mammal Foundation, 2240 Shelter Island Dr., San Diego, CA 92106, dorian.houser@nmmfoundation.org), James J. Finneran, Keith Jenkins, E. E. Henderson (US Navy Marine Mammal Program, San Diego, CA), Jason Mulsow, Brian K. Branstetter (National Marine Mammal Foundation, San Diego, CA), Patrick Moore (National Marine Mammal Foundation, Spring Valley, CA), and Stephen W. Martin (National Marine Mammal Foundation, San Diego, CA)

The United States Navy Marine Mammal Program (MMP) has been in existence for over 50 years. Following its inception, the program quickly became involved in the study of marine mammal sensory systems and bioacoustics. Early studies included the pioneering work of C. Scott Johnson in obtaining the first behavioral audiogram in a dolphin and Sam Ridgway's electrophysiological studies of dolphin hearing and sound production. Marine mammal bioacoustic studies grew substantially in the decades following the MMP's inception, and included numerous investigations into odontocete biosonar, pinniped and odontocete hearing (using both behavioral and physiological methods), and the impact of human-made sound on the hearing, behavior, and physiology of marine mammals. The MMP's bioacoustic research has significantly contributed to the Navy's environmental stewardship mandate (i.e. to predict and mitigate the impact of Navy activities on marine mammals), the development of bio-inspired sonar systems, and the ability to assess the hearing capabilities of marine mammals in the wild, under human care, and in stranded or rehabilitation scenarios. The MMP continues its bioacoustic studies today with investments focused on bottlenose dolphin and sea lion bioacoustics, but with expansion to the passive acoustic monitoring of wild marine mammals.

10:55

2aAB9. History of marine bioacoustics in Florida and the Caribbean. David Mann (Loggerhead Instruments, Inc., 6576 Palmer Park Circle, Sarasota, FL 34238, dmann@loggerhead.com)

The diverse fauna of Florida and the Caribbean has long attracted scientists studying marine bioacoustics. The earliest studies (1905–1945) in Florida examined species distributions and life histories. After World War II bioacoustics studies began in earnest. At Marineland, which opened to support underwater filming, Lilly studied dolphin vocalizations and attempted to link them to behavior. Melba and David Caldwell pioneered studies on the development of dolphin signature whistles. Tavolga performed fundamental work on sound production and behavior in gobies and many studies on the hearing ability of fishes at Bimini, Marineland, and Mote Marine Laboratory. Myrberg's Miami laboratory showed that different damselfishes had distinct sounds. In the 1960's, Breder documented diel and seasonal variation in fish sound production. Most recently, studies have focused on wild animals. Working with Wells' Sarasota Dolphin Research Project, Tyack, Sayigh, and Janik have cataloged signature whistles and used playback experiments to study salient features. Fundamental work identifying sounds produced by fishes begun by Lobel has spread to several research groups in Florida, USVI, Puerto Rico, and the Cayman Islands. Passive acoustic studies using gliders show that much work remains to identify the sources of unknown sounds.

11:15

2aAB10. History of bioacoustics research on aquatic and marine organisms in Hawaii. Whitlow Au (Univ. of Hawaii, P.O. Box 1346, Kaneohe, HI 96744, Kailua, HI 96734, wau@hawaii.edu)

Modern bioacoustic research with aquatic animals began in Hawaii in the 1968–1969 period when three independent programs started almost simultaneously. Two faculty members began studying various facets of hearing in fish at the University of Hawaii. Another faculty member from a different department established a laboratory in which dolphin hearing and echolocation were among some of the topics studied. The Navy started a dolphin facility to train dolphins to perform Navy related tasks. The Navy program also included research on dolphin echolocation including a study comparing the performance of two dolphins and a Straza-500 CTFM sonar in target detection. The Navy facility closed in 1993 but some of the bioacoustics research was transferred to the University's Hawaii Institute of Marine Biology on Coconut Island about a mile from the former Navy facility in the same bay. The HIMB program expanded into field research with spinner dolphins and humpback whales, and passive acoustic monitoring of marine organisms. The local National Marine Fisheries Service of NOAA became a close partner in the PAM development and studies. This presentation will focus of the bioacoustic research performed with marine mammals highlighting some of the more important findings and contributions to the field.

11:35

2aAB11. The central role of Woods Hole Oceanographic Institution (WHOI) in marine mammal bioacoustics. Douglas Wartzok (Dept. of Biological Sci., Florida Int. Univ., 11200 SW 8th St., Miami, FL 33199, wartzok@fiu.edu) and G. Carleton Ray (Environ. Sci., Univ. of Virginia, Charlottesville, VA)

WHOI researchers pioneered marine mammal bioacoustics. Schevill and Lawrence in 1949 made the first recordings of marine mammals, beluga whales, and in 1956 first demonstrated cetacean echolocation. Schevill later concentrated on the taxonomy and behavior of cetaceans while Watkins developed the first portable high-frequency recorder and passive and active acoustic tracking systems. Ray, Watkins, and Schevill in 1969 presented the first evidence of song in a marine mammal, the bearded seal, linked to the behavior of males in the breeding season. Watkins' papers on vocalizations informed a generation of bioacousticians on how to interpret sonograms. He was one of the first to use the Navy's SOSUS hydrophone array to track cetacean movements. Schevill and Watkins also inspired the research of others, both in recoding marine mammal sounds and in interpreting sound in behavior. Tyack led the development of the D-Tag, which simultaneously records vocalizations and movements of cetaceans. Ketten used CAT scans to develop hearing models for cetaceans. Sayigh advanced knowledge of delphnid signature whistles and Fristrup further developed the concept of soundscapes. Watkins and Schevill's database of more than 20,000 vocalizations from 70 marine mammal species now resides at the New Bedford Whaling Museum, freely available to the public.

TUESDAY MORNING, 8 MAY 2018 GREENWAY A, 8:00 A.M. TO 10:35 A.M.

Session 2aAOa

Acoustical Oceanography and Underwater Acoustics: Acoustic Seabed Characterization I

David P. Knobles, Cochair
KSA LLC, PO Box 27200, Austin, TX 78755

Preston S. Wilson, Cochair
Mech. Eng., Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292

Invited Papers

8:00

2aAOa1. Bayesian geoacoustic inversion for sediment profiles consisting of a general gradient over a layered substrate. Stan E. Dosso, Josée Belcourt (School of Earth & Ocean Sci, Univ of Victoria, PO Box 1700, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca), Jan Dettmer (GeoSci., Univ. of Calgary, Calgary, AB, Canada), and Charles W. Holland (Appl. Res. Lab., The Penn State Univ., State College, PA)

This paper considers Bayesian geoacoustic inversion and uncertainty estimation for seabed profiles consisting of a smooth gradient of general form overlying a potentially-layered substrate. The profile gradient is parameterized as a linear combination of Bernstein-polynomial basis functions, with the polynomial order determined objectively by applying the deviance information criterion. In this way the form of the gradient is determined by the data, rather than by a subjective prior choice. The substrate below the gradient consists

of uniform layers defined by an unknown number of (zero or more) interfaces, sampled probabilistically with trans-dimensional inversion. (Zero interfaces corresponds to a half-space with uniform parameter values equivalent to those at the base of the gradient.) The inversion approach is well suited to seabed environments consisting of an upper layer of soft, fine-grained sediments in which geoacoustic properties vary as a smooth function of depth overlying a harder, layered substrate, such as at the ONR Seabed Characterization Experiment at the New England mud patch. [Research funded by the Office of Naval Research, Ocean Acoustics Program.]

8:20

2aAOa2. Geoacoustic inversion of the New England Mud Patch using modal time-frequency dispersion. Julien Bonnel (Appl. Ocean Phys. and Eng. Dept., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS# 11, Woods Hole, MA 02543-1050, jbonnel@whoi.edu), Dimitrios Eleftherakis (LabSTICC, ENSTA Bretagne, Brest, France), Stan E. Dosso, N. Ross Chapman (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), David R. Dall'Osto, and Peter H. Dahl (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

This paper presents geoacoustic inversion of several low-frequency broadband signals recorded during the Seabed Characterization Experiment (SBCEX) that took place on the New England Mud Patch in March 2017. The considered sources are chirps emitted by a towed J15 source and underwater impulses created by MK64 explosions. These sources are low-frequency (f<250 Hz) so that the shallow water environment acts as a dispersive waveguide, and propagation is conveniently described by modal theory. The received signal is recorded on a single hydrophone placed 0.8 m off the bottom. In this context, inversion is carried out by matching modal dispersion curves in the time-frequency domain. Experimental dispersion curves are estimated using a non-linear sampling scheme called warping. Up to six modes can be estimated, and non-linear inversion results are consistent with what is known about the area.

8:40

2aAOa3. Inversions for sediment parameters using geophone and tetrahedral array measurements in the New England Mud Patch. Gopu R. Potty, James H Miller (Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882, potty@egr.uri.edu), Ying-Tsong Lin, Julien Bonnel, and Arthur Newhall (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Measurements of acoustic pressure and particle velocity were made during the Seabed Characterization Experiment (SCEx) in the New England Mud Patch south of Cape Cod in about 70 meters of water. The University of Rhode Island and Wood Hole Oceanographic Institution deployed the "geosled" with a four-element geophone array, a tetrahedral array of four hydrophones, and several hydrophone receive units (SHRUs). In addition, a new low frequency source, interface Wave Sediment Profiler (iWaSP) was deployed to excite interface waves (Scholte waves). The iWaSP system consists of a source to generate the interface wave and a four-element accelerometer receive array. Results of inversions for geoacoustic parameters will be presented. Arrival time dispersion from broadband sources (SUS and CSS) and iWaSP will be used for these inversions. Both p-wave and Scholte wave arrivals will be investigated using the geophone and hydrophone data. Seismic data collected at the location will be used to constrain the model parameters in the inversion. Sediment data from cores, other *in-situ* measurements and inversions using other types of data will be used to compare and validate our inversions. [Work supported by Office of Naval Research.]

9:00

2aAOa4. Attenuation in fine-grained sediments for New England Mudpatch in 25–5000 Hz band. David P. Knobles (KSA, LLC, PO Box 27200, Austin, TX 78755, dpknobles@yahoo.com), Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and William S. Hodgkiss (UC San Diego, La Jolla, CA)

A subject of considerable importance in littoral ocean environments is the frequency dependence of sediment attenuation for various classes of sediments. The well-known frequency exponent value of 1.8 for sandy sediments was the subject of numerous research studies. For mud-like or mixed sediments, for which there is a scarcity of experimental data, values for the frequency exponent are not well established. Previously, the sound speed structure of a surface mud layer in the New England Seabed Characterization experimental area was estimated with inversion and Bayesian methods. The inversions utilized MK-64 SUS explosive sources in a 25-275 Hz band placed on concentric circles of 2, 4, and 6.5 km radii in the latitude-longitude plane. The area was previously surveyed with CHIRP sonar allowing for multiple sediment layer horizons to be constrained for layer thicknesses by measured two-way travel times. In this analysis, a larger bandwidth (25-5000 Hz) of the SUS, the Combustive Sound Source (CSS), and tonals from towed sources at multiple frequencies allow for the development of posterior probability distributions that contain the statistics of the frequency dependence of sediment attenuation up to about 5 kHz. [Work supported by ONR]

9:20–9:35 Break

Contributed Papers

9:35

2aAOa5. Source characteristics and variability of impulsive sources used in ocean acoustics experiments. Matthew C. Zeh, Preston S. Wilson, Andrew R. McNeese (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 3374 Lake Austin Blvd., Apt. B, Austin, TX 78703, mzeh@utexas.edu), and David P. Knobles (Knobles Sci. and Anal., LLC, Austin, TX)

Two types of impulsive sources were used extensively in the Seabed Characterization Experiment 2017 (SCE2017): Mk-64 Sound Underwater Source (SUS), and the Combustive Sound Source (CSS). Both of these sources rely on the release of chemical energy and hence the radiated acoustic signal is subject to variability from shot to shot. In addition, SUS activates via a pressure sensor and there is variability in the depth of activation, which translates into additional variability in the radiated acoustic signal. A pair of ship-deployed hydrophones was used during SCE2017 to record source waveforms, and in this presentation, various acoustic characteristics of these sources, such as source level, frequency content, and the nature of the time domain signals will be described in terms of their statistical variability. [Work supported by ONR.]

9:50

2aAOa6. Distinctions among physical types of marine sediments, especially mud, and their possible fundamentals-based and heuristic models. Allan D. Pierce (Cape Cod Inst. for Sci. and Eng., PO 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)

Marine sediments at different locations and depths are not necessarily similar. Distinctions are (1) sandy/silty sediments, when most particles have settled and almost every particle touches other particles, and (2) mud sediments, where the majority in a layer are suspended so that they do not touch other particles. Most theoretical and analytical modeling over the years has largely been confined to sandy/silty sediments, and some successful models, although arguably heuristic and with adjustable parameters, have emerged. There is also some speculation that such models might apply equally well for mud sediments. The present authors in recent years have concentrated on the development of a fundamentals-based model for mud sediments, although there is as yet insufficient data to adequately test its validity. The present paper speculates on what this model may have in common with models for sandy/silty sediments. There are expectations that the Mallock-Wood model may approximately apply for sound speed predictions at very low frequencies. Arguments are presented to the effect that the physics is sufficiently different for the two sediments types that a substantially different model is needed for mud sediments. One chief distinction is that the solid portion of mud sediments has a portion that is clay particles, and these have substantially different electrolytic and cohesive properties than those of quartz particles.

10:05

2aAOa7. Acoustics of biologically active marine sediments. Kevin M. Lee, Megan S. Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, klee@arlut.utexas.edu), Gabriel R. Venegas, and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Biological organisms are prevalent on and near the seabed; however, most predictive theoretical models do not explicitly address the acoustic behavior of biologically active marine sediments. Sediment-acoustics models, such as those based on Biot theory or grain-shearing theory, allow for parameterization of various physical properties including bulk density, porosity, shear strength, or pore fluid viscosity, all of which can be modified by the presence of biological organisms or bioturbation. Other models can include the effects gas bubble distributions, which can also be associated with the presence of benthic biology, on compressional and shear wave propagation. In this paper, we examine sediment-acoustics models in comparison with data from various field experiments focused on infauna- and seagrass-bearing sediments. The model parameters are partially constrained by physical data from core samples collected at the field experiment sites. Empirical relations between the core data parameters and acoustic data will also be examined. [Work supported by ONR and ARL:UT IR&D.]

10:20

2aAOa8. Implications from a clay/silt suspension model of mud for data from the Seabed Characterization Experiment. Elisabeth M. Brown (Mathematical Sci., Rensselaer Polytechnic Inst., Rensselaer Polytechnic Inst., Dept Mathematical Sci., Troy, NY 12180, browne6@rpi.edu), Allan D. Pierce (Cape Cod Inst. for Sci. and Eng., East Sandwich, MA), Ying-Tsong Lin (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), Jason D. Chaytor (Coastal and Marine Sci. Ctr., U. S. Geological Survey, Woods Hole, MA), and William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

Mud-like marine sediments may be modeled as a suspension of flocculated clay particles in water, in which silt particles are embedded in the clay flocs. Recent calculations based on this model [Pierce *et al.*, New Orleans ASA] produced new predictions for the frequency-dependent phase speed and attenuation of compressional waves. For phase speed, very good comparisons between model predictions and archival data were found. Comparisons will be presented using geoacoustic data from sediment cores extracted from the New England Mud Patch. For attenuation, predictions using Gaussian distributions of silt particle radius were calculated. An approach will be described for producing a frequency-dependent effective particle radius to account for size-distribution effects using single-particle attenuation expressions. The influence of non-Gaussian size distributions on the effective radius and on the frequency intervals of near-linear attenuation behavior will be shown. An environmental model for a WHOI AUV track deployed in the ONR Seabed Characterization Experiment was constructed, and initial comparisons were made between transmission loss calculations and acoustic data. Benchmarking will be performed with additional forward calculations, along with estimates of geoacoustic properties. Particular interest is in sediment attenuation and its frequency dependence. [Work supported by ONR.]

Session 2aAOb

Acoustical Oceanography: Acoustical Oceanography Prize Lecture

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

Chair's Introduction—11:00

Invited Paper

11:05

2aAOb1. Three-dimensional shallow water sound propagation and applications toward acoustical oceanography. Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu)

Underwater sound propagation in areas of the continental shelf, shelf break, and continental slope can encounter strong horizontal reflection, refraction, diffraction, focusing, and/or defocusing due to a variety of environmental factors, including bathymetric variability, water column fluctuations (internal waves and shelf-break fronts), and boundary roughness. Theoretical, numerical, and experimental approaches have been taken to investigate the underlying physics of these three-dimensional (3-D) sound propagation effects and their temporal and spatial variability induced jointly by marine geological features and dynamic oceanographic processes. Some recent work on theoretical analysis, numerical modeling and field work experiments to study 3-D sound propagation in nonlinear internal wave ducts, over the continental slope and in submarine canyons will be reviewed in this talk. The ultimate goal of investing these 3-D sound propagation effects is to improve long-distance acoustical oceanographic technology in complex ocean environments. An example using a 3-D back-propagation method for acoustic inverse problems will be presented, along with discussion on feature applications. [Work supported by the Office of Naval Research.]

Session 2aBA

Biomedical Acoustics: Using Acoustic Wave Propagation to Estimate Quantitative Material Properties of Tissue I

Matthew W. Urban, Chair
Department of Physiology and Biomedical Engineering, Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905

Chair's Introduction—8:00

Invited Papers

8:05

2aBA1. Magnetic resonance elastography: Status of clinical applications. Richard L. Ehman (Dept. of Radiology, Mayo Clinic, 200, Rochester, MN 55905, ehman.richard@mayo.edu)

Magnetic resonance elastography (MRE) is an imaging technology that has been available as an FDA-cleared upgrade to MRI systems since 2009. Worldwide, MRE technology has been installed on approximately 1000 MRI systems. In October 2017, the AMA approved the creation of a CPT code for MRE. In MRE, propagating shear waves in the range of 30–300 Hz are generated in tissue, and imaged with a phase contrast MRI technique. The wave images are processed with an inversion algorithm to generate cross-sectional images quantitatively depicting the complex shear modulus of tissue. The main application of MRE is currently in noninvasive diagnosis of chronic liver disease. Published studies have established the MRE is the most accurate and technically successful non-invasive modality for diagnosing liver fibrosis. Many other promising applications of MRE are being explored by investigators around the world, including assessment of diseases of the brain, heart, lung, and muscle and tumors of breast, liver, prostate, liver, and other organs. The new applications that closest adoption in clinical practice are (1) assessment of brain disease and (2) advanced diagnosis of liver disease. These new applications require acquisition sequences and processing algorithms that are more advanced than the current product MRE techniques.

8:25

2aBA2. Magnetic resonance elastography inversions: Technical challenges and recent developments. Armando Manduca (Physiol. and Biomedical Eng., Mayo Clinic, Opus 2-125, Mayo Clinic, 200 1st St. SW, Rochester, MN 55901, manduca@mayo.edu)

Magnetic resonance elastography (MRE) is a phase contrast based MRI imaging technique that can quantitatively and non-invasively measure full 3D vector displacement data from propagating acoustic waves *in vivo*. The data acquired allow the calculation of local values of complex shear modulus and the generation of images that depict the viscoelastic properties of tissue. Acquisitions at different frequencies can capture the dispersive behavior of these quantities, and advanced approaches attempt to map anisotropic and poroelastic properties of materials. We will discuss current approaches for the inversion of MRE data, focusing on the issues and assumptions involved, and highlight some of the challenges faced in such inversions, particularly in thin or small structures in which wave propagation is dominated by waveguide effects. We will also present results based on neural network inversions, which may have lower repeatability error and be more resistant to noise than some current approaches.

8:45

2aBA3. Spectral-based quantitative ultrasound imaging: A model free approach. Michael L. Oelze, Trong Nguyen, and Minh Do (Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 405 N Mathews, Urbana, IL 61801, oelze@uiuc.edu)

Quantitative ultrasound (QUS) imaging can improve the diagnostic capabilities of ultrasound. Spectral-based QUS approaches utilize backscattered ultrasound signals to extract additional information about tissue state. These technique have traditionally relied on scattering models to differentiate tissue state. However, the use of machine learning approaches may obviate the need for models. A model-based and model-free (principle component analysis (PCA)) approach to spectral-based QUS were compared for their ability to differentiate fatty from non-fatty liver in a rabbit model. PCA was observed to provide better differentiation of fatty from non-fatty liver, i.e., PCA predicted fatty liver 86% of the time versus 36% for model-based. In addition, three calibration approaches for spectral-based QUS were compared: the traditional reference phantom, reference free, and an *in situ* calibration target. To test the calibration procedures, a phantom was scanned with and without a lossy layer placed on top and integrated backscatter coefficients (IBSCs) were calculated from the scattered data. Utilizing an *in situ* calibration target provided the ability to account for transmission losses and attenuation. The root mean square error between IBSCs estimated from the phantom with and without the lossy layer were 8.28 for the reference phantom versus 1.24 for the *in situ* calibration approach.

9:05

2aBA4. Fast simulations of shear waves generated by an acoustic radiation force: Comparisons with measurements in shear wave phantoms. Yiqun Yang (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI), Matthew W. Urban (Dept. of Radiology, Mayo Clinic College of Medicine and Sci., Rochester, MN), and Robert J. McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI, mcgough@msu.edu)

We recently developed a fast approach for simulating shear waves generated by an acoustic radiation force. The input for these simulations is generated in FOCUS, the "Fast object-oriented C++ simulator" (www.egr.msu.edu/~fultras-web), which rapidly models the three-dimensional (3D) pressure, intensity, and acoustic radiation force on a desktop computer. The shear waves generated by the acoustic radiation force are then quickly simulated with Green's functions for viscoelastic media in a two-dimensional (2D) plane on a high-performance graphics processing unit (GPU). For these simulations, an L7-4 linear array is electronically focused at a depth of 25 mm to generate an acoustic radiation force "push" for 200 microseconds. The simulation results are compared to shear wave data measured in three viscoelastic shear wave phantoms with low, medium, and high values of the shear viscosity using a Verasonics Vantage ultrasound system. The measured and simulated shear wave data are compared for several different combinations of the simulation parameters, where these comparisons are enabled by the rapid acoustic radiation force and shear wave simulations. The results show that the shear waves simulated in 2D planes achieve good agreement with the measured shear wave data. [Work supported in part by NIH Grants EB023051 and DK092255.]

9:25

2aBA5. Efficient computational algorithms for modeling guided wave dispersion in arterial walls. Ali Vaziri (MSC Software Corp., Raleigh, North Carolina), Matthew W. Urban (Mayo Clinic, Rochester, MN), Wilkins Aquino (Duke Univ., Durham, NC), James F. Greenleaf (Mayo Clinic, Rochester, MN), and Murthy Guddati (NC State Univ., 2501 Stinson Dr., NCSU-Civil Eng., Raleigh, NC 27695-7908, mnguddat@ncsu.edu)

Arterial stiffness is a well-known biomarker of early cardiovascular disease. Shear wave dispersion ultrasound vibrometry (SDUV) has emerged as a promising technique to estimate local arterial stiffness from the observed dispersion of guided waves. With the ultimate goal of developing real-time inversion for arterial stiffness from SDUV measurements, we develop and validate highly efficient and accurate computational algorithms that compute the wave dispersion in multi-layered immersed tubes. The proposed approach carefully combines Fourier transformation and *one-dimensional* finite-element discretization to accurately capture fully *three-dimensional* wave propagation. The method is several orders of magnitude more efficient than three-dimensional finite element simulation, and eliminates other complexities such as the need for absorbing boundary conditions. The method is validated using SDUV experiments on tissue-mimicking phantoms. The validation exercise uncovered an important detail that is often overlooked—the dispersion curve captured through SDUV experiments does not correspond to a single dispersion curve, but a combination of multiple dispersion curves. This implies that proper identification of the dispersion curves could be critical to estimating the arterial stiffness. In this talk, we present the details of the proposed method including formulation, computational complexity, verification and validation.

9:45

2aBA6. Speckle-free estimation of tissue elasticity with single track location shear wave elasticity imaging. Peter J. Hollender (Biomedical Eng., Duke Univ., Rm. 1427, FCIEMAS Box 90281, 101 Sci. Dr., Durham, NC 27708, peter.hollender@duke.edu)

Shear Wave Elasticity Imaging (SWEI) is commonly used to characterize tissue elasticity, but conventional, multiple-track-location SWEI (MTL-SWEI) techniques are resolution-limited by speckle. MTL-SWEI techniques use plane wave ultrasound to monitor an induced shear wave as it propagates across a set of tracking beams within a region of interest. The scattering process creates a random, yet stationary, spatial sensitivity pattern for each beamformed location, called speckle bias, which causes errors in MTL-SWEI shear wave speed estimates that cannot be improved through averaging. Single Track Location SWEI (STL-SWEI) overcomes speckle bias by using a single track beam, subsequently exciting and tracking different push locations, and comparing the timing of the recorded shear wave signals from different push locations to estimate shear wave speed. Two and three-dimensional STL-SWEI imaging techniques are presented and compared to MTL-SWEI and Acoustic Radiation Force Impulse (ARFI) imaging in phantoms and *in vivo* experiments. Tradeoffs and techniques for sequencing, beamforming, shear wave speed estimation, and image formation are discussed. For applications where tissue heating and motion are not limiting factors, STL-SWEI provides superior imaging in terms of lateral resolution and contrast-to-noise ratio compared to MTL-SWEI and ARFI. [This work was supported by NIHR37HL096023 and NIHR01EB01248.]

10:05–10:20 Break

10:20

2aBA7. Identifying nonalcoholic fatty liver disease from quantitative ultrasound measures. W. D. O'Brien, Jr., Aiguo Han (BioAcoust. Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 306 N. Wright St., Urbana, IL 61801, wdo@uiuc.edu), Ethan Z. Sy (Dept. of Radiology, Univ. of California at San Diego, La Jolla, CA), Rohit Loomba (Dept. of Medicine, Univ. of California at San Diego, La Jolla, CA), Claude B. Sirlin (Dept. of Radiology, 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), and Michael P. Andre (Dept. of Radiology and the San Diego VA Healthcare System, Univ. of California at San Diego, La Jolla, CA)

Nonalcoholic fatty liver disease (NAFLD) is the most common cause of chronic liver disease in the United States, affects 30% of adult Americans, may progress to nonalcoholic steatohepatitis (NASH) and end-stage liver disease, and is a risk factor for diabetes and cardiovascular disease. The diagnosis, grading, and staging of NAFLD currently is based on liver biopsy with histologic analysis. Noninvasive image-based methods to evaluate the liver in adults with NAFLD are urgently needed. The objective is to identify the relationships between quantitative ultrasound (QUS) outcomes [backscatter coefficient, BSC, and attenuation coefficient, AC] and tissue properties to validate the diagnosis and/or grading of NAFLD. 83 participants with known or suspected NAFLD (thus few uncompromised liver samples expected) received contemporary QUS and MRI (proton density fat fraction) PDFF to estimate liver fat fraction. Of this group, 74 participants also received liver biopsy. Participant recruitment is continuing. Currently, observations show good correlations of both BSC and AC with PDFF, where PDFF ranged from 3% to 42%. Of the 74 biopsy participants, all stages of fibrosis have been pathologically identified [F0,25; F1,22; F2,13; F3,10; F4,4] suggesting both QUS identification opportunities and confounder challenges. [Support: NIH R01DK106419 and NIH R37EB002641.]

10:35

2aBA8. A nonlinear grid-search inversion for cortical bone thickness and ultrasonic velocities. Tho N. Tran (Dept. of Radiology and Diagnostic Imaging, Univ. of AB, Res. Transition Facility, Edmonton, AB T6G 2V2, Canada, tho.tran@ualberta.ca), Mauricio D. Sacchi (Dept. of Phys., Univ. of AB, Edmonton, AB, Canada), Vu-Hieu Nguyen (Laboratoire Modélisation et Simulation Multi Echelle UMR 8208 CNRS, Université Paris-Est, Creteil, France), Dean Ta (Dept. of Electron. Eng., Fudan Univ., Shanghai, China), and Lawrence H. Le (Dept. of Radiology and Diagnostic Imaging, Univ. of AB, Edmonton, AB, Canada)

Cortical thickness and elasticity are important determinants of bone strength. This study proposes a nonlinear grid-search inversion to estimate the thickness and ultrasonic velocities of long cortical bones from axially-transmitted data. The inversion scheme has been developed in the frequency-phase velocity domain to recover bone properties. The method uses ultrasonic guided waves to retrieve cortical thickness, compressional, and shear-wave velocities of the cortex. The inversion strategy requires to systematically examine a set of trial dispersion curves within a pre-defined model space to match the data which minimizes the objective function in a least-squares sense. The theoretical dispersion curves required by the inversion are computed for bilayered bone models using a semi-analytical finite-element method. Its application demonstrates the feasibility of the proposed approach on synthetic data for a 5 mm-thick bone plate, and *in-vitro* dataset from a bovine femur plate with 2 mm-thick soft-tissue mimic on top. Our results indicate that one can recover the cortical thickness and wave speeds with less than 5% error. The accuracy can be improved by refining the grid; however, the computational cost of the method will increase. In the future, we aim at applying the method to clinical data.

10:50

2aBA9. Characterization of ultrasound attenuation and angular distribution from trabecular bone phantom material using refracto-vibrometry. Benjamin A. Rorem, Matthew R. Mehrkens, Thomas M. Huber (Phys., Gustavus Adolphus College, 800 W College Ave., Saint Peter, MN 56082, huber@gac.edu), Matthew T. Huber, and Brent Hoffmeister (Phys., Rhodes College, Memphis, TN)

Ultrasound is being researched and used for diagnosing osteoporosis. Researchers use trabecular bone from humans as a test material, but such bone is bio-hazardous, not uniform, and of limited size. Recently it has been demonstrated that the open-cell polyurethane foam, known as "Sawbones" could be utilized as a substitute for trabecular bone in ultrasound studies. In the current study, rectangular slices of Sawbones trabecular bone phantom material were insonified with a 25 mm diameter 500 kHz ultrasound transducer oriented both normal to the surface and at an angle. Refracto-vibrometry (RV), an interferometric method for optically measuring ultrasound, was compared with conventional transducer measurements of ultrasound transmission through the bone phantom samples. The measurement beam from a Polytec PSV-400 scanning laser Doppler vibrometer was directed through a water tank towards a stationary retroreflective surface. Acoustic wave fronts (density variations) which pass through the ~50μm diameter measurement laser cause variations in the integrated optical path length. The measured signals were used to determine parameters such as normalized broadband ultrasonic attenuation (nBUA) at numerous scan points. This enabled measurements of the spatial and angular distribution of the transmitted ultrasonic field through the sample that are not possible using a conventional single-element ultrasonic transducer.

11:05

2aBA10. Characterizing anisotropic, viscoelastic material properties of the tectorial membranes of wild-type and mutant mice using longitudinally propagating waves. Charlsie Lemons (G.W.W. School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332, charlsielem@gmail.com), Jonathan B. Sellon, Roberto D. Ortiz (Res. Lab. of Electronics, Massachusetts Inst. of Technol., Cambridge, MA), Elisa Boatti (G.W.W. School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Dennis M. Freeman (Res. Lab. of Electronics, Massachusetts Inst. of Technol., Cambridge, MA), and Julien Meaud (G.W.W. School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

The tectorial membrane (TM) is an important structural component of the mammalian inner ear. Examination of cochlear physiology in genetically modified mice has demonstrated that mutation of genes affecting TM proteins causes changes in key characteristics of cochlear function. Characterizing the differences in material properties between wild-type and mutant mice could provide insight into the source of reported differences in cochlear physiology. In this study, optical images of isolated TM segments of wild-type and genetically modified mice in response to harmonic radial excitation at acoustic frequencies are used to determine the amplitude and phase of the motion. Wave propagation on the TM segments is modeled using finite element models that take into account the anisotropy, viscoelasticity, and finite dimensions of the TM and the presence of a viscous boundary layer. An automated least-square fitting algorithm is used to find anisotropic, viscoelastic material parameters of wild-type and mutant mice TMs at acoustic frequencies. The resulting material properties are compared to previous estimates of the TM properties.

11:20

2aBA11. Lung ultrasound surface wave elastography for assessing interstitial lung disease. Xiaoming Zhang, Boran Zhou, Ryan Clay, Jinling Zhou, Thomas Osborn, Brian Bartholmai, James F. Greenleaf, and Sanjay Kalra (Mayo Clinic, 200 1st ST SW, Rochester, MN 55905, zhang.xiaoming@mayo.edu)

A lung ultrasound surface wave elastography (LUSWE) technique is developed for assessing interstitial lung disease (ILD). In LUSWE, a 0.1 second harmonic vibration is generated on the chest wall of a subject using a handheld vibrator. An ultrasound probe is aligned with the vibration indenter in the same intercostal region to measure the generated surface wave speed of the lung. A human subject is examined in a sitting position. The lung is tested at the total lung volume. The upper anterior lungs are tested through the second intercostal space in the mid-clavicular line. The lower lateral lungs are tested in the mid-axillary line and the lower posterior lungs are tested in the mid-scapular line. In a prospective clinical study, we measure both lungs through the six intercostal regions for patients and healthy controls. The surface wave speed is measured at 100 Hz, 150 Hz, and 200 Hz. Significant differences of wave speed between patients and controls were found in all lung regions at all frequencies. We also found positive correlation between LUSWE and CT analyses. LUSWE is a safe and noninvasive technique for generating and measuring surface wave propagation on the lung. LUSWE may be useful for assessing ILD.

11:35

2aBA12. Fingertip elastography. Altaf Khan (Mech. Eng., Univ. of Illinois at Chicago, Chicago, IL) and Thomas J. Royston (BioEng., Univ. of Illinois at Chicago, 851 South Morgan St., MC 063, Chicago, IL 60607, troyston@uic.edu)

Dynamic elastography methods attempt to quantitatively map soft tissue viscoelastic properties. Application to the fingertip, relevant to medical diagnostics and to improving tactile interfaces, is a novel and challenging application, given the small target size. In this feasibility study, an annular actuator placed on the surface of the fingertip and driven harmonically at multiple frequencies sequentially creates geometrically focused surface waves. These surface wave propagation patterns are measured using scanning laser Doppler vibrometry. Reconstruction (the inverse problem) is performed in order to estimate fingertip soft tissue viscoelastic properties. The study identifies limitations of an analytical approach versus an optimization approach that utilizes a finite element model. Measurement at multiple frequencies reveals limitations of an assumption of homogeneity of material properties. Identified shear viscoelastic properties increase significantly as frequency increases and the depth of penetration of the surface wave is reduced, indicating that the fingertip is significantly stiffer near its surface. Additional studies using Optical Coherence Tomography (OCT) and Magnetic Resonance Imaging (MRI) provide insights into wave propagation at near sub-surface and deep sub-surface zones.

TUESDAY MORNING, 8 MAY 2018 GREENWAY D, 9:00 A.M. TO 12:00 NOON

Session 2aEA

Engineering Acoustics and Physical Acoustics: Miniature Acoustic Transducers I

Vahid Naderyan, Chair
Physics/National Center for Physical Acoustics, University of Mississippi, NCPA, 1 Coliseum Drive, University, MS 38677

Chair's Introduction—9:00

Invited Papers

9:05

2aEA1. A revisit to the lumped element model of capacitive MEMS microphones. Yu Du, Michael Kuntzman, and Yenhao Chen (Knowles Corp., 1151 Maplewood Dr., Itasca, IL 60143, Yu.Du@knowles.com)

Lumped element model (LEM) in the format of equivalent circuits are commonly used to describe dynamic characteristics of electromechanical transducers such as capacitive MEMS microphones. The derivation of such a LEM relies on classical electroacoustic and electromechanical principals and assumes physical elements with simple geometries and boundary conditions. However, practical designs rarely meet these assumptions, which leads to difficulties to extract model parameters with pure theoretical methods and limits the model usage to guide real designs. This study presents an improved LEM for MEMS microphones in which a hybrid theoretical-experimental method is used to extract model parameters. In particular, the mechanical compliance of the moving diaphragm and the transduction coefficient from the mechanical domain to the electrical domain are directly measured using laser Doppler vibrometer, which accounts for the influences of complex boundary conditions and geometrical shapes of actual designs. Furthermore, the diaphragm softening effect due to capacitive loading in the interface circuits is also included. Results predicted by the improved LEM are compared to measurements. It is shown that the newly proposed model can accurately capture the microphone response in the audio region. With care, its use may be extended to the ultrasonic range up to 40–50 kHz.

9:25

2aEA2. Multidomain modeling of acoustic transducers using SimScape. Thomas Blanford (Graduate Program in Acoust., The Penn State Univ., 1623 S. Ashwicken Ct, State College, PA 16801, teb217@psu.edu) and Stephen C. Thompson (Graduate Program in Acoust., The Penn State Univ., University Park, PA)

Finite element modeling of acoustic transducers can capture many complex behaviors that are neglected in simpler analyses. However, the complexity that requires finite element modeling is often confined to a small region of the total device. In that case it is possible to combine the finite element model of the subsection that requires this complexity with dynamic system modeling methods for the balance of the design. Dynamic system modeling includes earlier methods of analog circuit modeling and generalizes those methods appropriately tor acoustic transducer analysis. This paper will describe SimScape modeling of transducer structures in a way that allows combined modeling with FEA where needed. Simple examples will be selected from among moving coil speakers, balanced armature transducers, condenser microphones, and piezoelectric devices.

9:45

2aEA3. Biologically inspired acoustic sensors: From insect ears to miniature microphones. James F. Windmill (Dept. of Electron. & Elec. Eng., Univ. of Strathclyde, 204 George St., Glasgow G1 1XW, United Kingdom, james.windmill@strath.ac.uk)

Taking inspiration from insect ears to develop an acoustic device is not new. Research to take inspiration from the highly directional, sub-wavelength, *Ormia ochracea* fly's ear to produce miniature directional microphones is well-known. Since the 1990s, researchers have tried to implement *Ormia* based micro-electro-mechanical systems devices, typically using standard silicon or similar microfabrication. Much of this time has been spent trying to circumvent the fact that *Ormia* evolved to hear one specific frequency, that of a calling cricket, and so is not a broadband system as you would require for an audio microphone. Further, building a silicon system inspired by the mechanics of insect ears leads to various compromises. This talk will discuss ongoing work at the University of Strathclyde on silicon microfabrication of acoustic devices inspired by the *Ormia* system. It will focus on efforts to create MEMS microphones designed with multiple resonance frequencies in the human vocal range. It will also describe ongoing research into the use of active feedback in MEMS acoustic systems, inspired by auditory hearing mechanisms. Finally, it will also look at the latest research by the Strathclyde team into the application of 3D microfabrication for the development of bio-inspired acoustic devices.

10:05

2aEA4. Miniature microphones: Non-standard fabrication approaches and associated modeling issues. Libor Rufer (TIMA Lab., Univ. Grenoble Alpes, France, 46, Av. Felix Viallet, Grenoble 38031, France, Libor.Rufer@univ-grenoble-alpes.fr)

Micro-electro-mechanical systems (MEMS) allowed miniature microphones that during the last few decades have left research laboratories to enter a massive industrial development. Their characteristics meet the most demanding specifications of various portable devices. This talk will show some non-standard approaches to the fabrication technology and associated modeling issues using two cases of specific designs in support. The first case is a piezoresistive microphone devoted to high-intensity and high-frequency acoustic fields. To damp the resonance, an atypical diaphragm structure was employed. Such a solution, allowing relatively large airgap required by large diaphragm displacements, will be presented and analyzed. The second case concerns a design of a microphone fabricated with a standard complementary metal oxide semiconductor (CMOS) technology. This technology, in contrast with dedicated technologies typically used by the microphone industry, is readily available to small companies and academic institutions. One of the drawbacks of the CMOS-MEMS approach is a relatively high degree of design constraints represented by materials and layers dimensions involved in the CMOS process. Moreover, the device layout must take into account specific requirements of the fabrication technique. This talk will present some of these constraints and their impact on the microphone performance that will be supported by simulation results.

10:25–10:40 Break

10:40

2aEA5. Spin-MEMS microphone based on highly sensitive spintronic strain-gauge sensors. Yoshihiko Fuji, Yoshihiro Higashi, Shiori Kaji (Corporate Res. & Development Ctr., Toshiba Corp., 1, Komukai-Toshiba-cho, Saiwai-ku, Kawasaki, Kanagawa 211-8583, Japan, yoshihiko.fuji@toshiba.co.jp), Kei Masunishi (Corporate Manufacturing Eng. Ctr., Toshiba Corp., Yokohama, Kanagawa, Japan), Tomohiko Nagata, Akiko Yuzawa, Kenji Otsu, Kazuaki Okamoto, Shotaro Baba, Tomio Ono, and Michiko Hara (Corporate Res. & Development Ctr. (Retired), Toshiba Corp., Kawasaki, Kanagawa, Japan)

We report a novel spintronic MEMS (Spin-MEMS) microphone, which is a new type of resistive microphone. For this microphone, spintronic strain-gauge sensors (Spin-SGSs) are integrated on a bulk micromachined diaphragm. The Spin-SGSs are based on magnetic tunnel junctions (MTJs) similar to those used as magnetic sensors in hard disk drives. In work to date, we have experimentally confirmed that the Spin-SGS exhibits a high gauge factor in excess of 5000, which is 100-fold that for a conventional poly-Si piezoresistor, by adopting a novel amorphous Fe-B-based sensing layer with high magnetostriction and low coercivity. Thanks to the high strain sensitivity of the Spin-SGSs, the Spin-MEMS microphone exhibits a signal-to-noise ratio (SNR) of 57 dB(A). A Spin-MEMS microphone with a first resonance frequency of over 70 kHz was also fabricated that exhibits an SNR of 49 dB(A), which is promising for acoustic health monitoring. In this study, we compared the operation sounds of defective and normal bearings using the Spin-MEMS microphone. The Spin-MEMS microphone detected differences in the operation sounds between the defective and normal bearings in the high-frequency range of 10 kHz to 50 kHz.

11:00

2aEA6. A MEMS condenser microphone based acoustic receiver for totally implantable cochlear implants. Flurin Pfiffner (Dept. of Otorhinolaryngology, Head and Neck Surgery, Univ. of Zurich, Univ. Hospital Zurich, Zurich, Zurich, Switzerland), Lukas Prochazka, Ivo Dobrev, Adrian Dalbert, Jae Hoon Sim (Dept. of Otorhinolaryngology, Head and Neck Surgery, Univ. of Zurich, Univ. Hospital Zurich, UniversitätsSpital Zürich, Frauenklinikstrasse 24, ORL Klinik, Zurich, Switzerland, Lukas.Prochazka@usz.ch), Francesca Harris (Cochlear Technol. Ctr., Mechelen, Belgium), Jeremie Guignard (Cochlear AG, Boston, Massachusetts), Joris Walraevens (Cochlear Technol. Ctr., Mechelen, Belgium), Christof Röösli, and Alex Huber (Dept. of Otorhinolaryngology, Head and Neck Surgery, Univ. of Zurich, Univ. Hospital Zurich, Zurich, Switzerland)

The goal of the present project is to develop intracochlear acoustic receivers (ICAR's) for measurement of the sound pressure in the inner ear of human temporal bones and in acute large animal experiments. In addition, the ICAR is designed to be used as an implantable microphone for totally implantable cochlear implant (TICI) systems. The presented ICAR concept is based on a commercially available MEMS condenser microphone customized with a protective diaphragm providing sealing properties and optimized sensor head geometry for accessing the tiny fluid-filled cavities of the human inner ear. The first ICAR prototypes (PT I) have been used for numerous intracochlear sound pressure measurements in human and sheep temporal bones. The data thus obtained are in good agreement with the literature. The second ICAR prototype (PT II) was further adapted for surgical insertion in the scala tympani in acute large animal experiments. First experiments have been successfully performed and further revealed that the presented ICAR concept is a suitable receiver technology for TICI systems. Currently, the development of a fully biocompatible ICAR (PT III) is ongoing. This sensor must fulfill all important requirements of a TICI device such as high performance, low power consumption and good system integration.

11:20

2aEA7. Survey of simulations techniques used in balanced armature transducers. Charles King (R&D, Knowles Electronics, 1151 Maplewood Dr, Itasca, IL 60302, charles.king@knowles.com)

Balanced armature transducers, also known as receivers, are commonly used in hearing aids, hearables, and in-ear monitors. These are difficult to simulate. They are very small devices, designed to fit inside the ear canal, and measurement of basic quantities is physically difficult. Additionally, multiple magnetic non-linearities are balanced to create a high-efficiency / high-output device. Spice simulations are well suited to understand the linear frequency response of the device. Finite element simulations are well suited to study steady state magnetic and geometric non-linearities. Time domain algebraic differential equation techniques can bridge these two simulation regimes. With this tool the amplitude response, frequency response, and distortion caused by the non-linearity's can be studied. The talk will examine strengths, weaknesses, and when it is appropriate to use each of these individual techniques.

11:40

2aEA8. Vibroacoustic simulation of hearing aid receivers. Brenno Varanda (Knowles Corp., 1410 Alexander Way, Clearwater, Florida 33756, bvarand1@gmail.com)

The overall aim of this research is to develop practical vibroacoustic models of hearing aid receivers, a key electro-acoustic component of hearing aids. The receiver is a high efficiency miniature sound source which utilizes a balanced armature electromagnetic motor. A standard side effect for most balance armature receivers is structural vibration. This receiver-borne structural vibration can travel through the hearing aid package to the microphones, resulting in undesirable oscillations, just like acoustic feedback. The receiver models are used to help hearing aid designers refine vibration isolation mounts and package components to reduce both acoustic and receiver-borne structural feedback. The model consists of a simplified electro acoustic circuit-equivalent that can easily be coupled to multi-physics finite element analyses. The model has been validated against standard hearing industry measurements and proved to be effective on predicting transmissibility forces across various speaker attachments.

Session 2aNS

Noise, Psychological and Physiological Acoustics, and Speech Communication: Hearing Health Across a Lifespan: Hearing Screening From Cradle to Grave

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

Alexander L. Francis, Cochair
Purdue University, SLHS, Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907

Chair's Introduction—8:30

Invited Papers

8:35

2aNS1. Hearing health across a lifespan: Hearing screening from cradle to grave. John Eichwald (National Ctr. for Environ. Health, Centers for Disease Control and Prevention (CDC), 1600 Clifton Rd. NE, MS-E18, Atlanta, GA 30333, jeichwald@cdc.gov)

The 2016 National Academies of Sciences, Engineering, and Medicine report "Hearing Health Care for Adults: Priorities for Improving Access and Affordability" included a call to action for government agencies to strengthen efforts to collect, analyze, and disseminate population-based data on hearing loss in adults. In partial response, the Centers for Disease Control and Prevention (CDC) analyzed the most recent available data collected both by questionnaire and audiometric tests of participants aged 20–69 years in the 2011–2012 National Health and Nutrition Examination Survey to determine the presence of audiometric notches indicative of noise-induced hearing loss. Prevalence of both unilateral and bilateral audiometric notches and their association with self-reported exposure to loud noise were calculated. Nearly one in four adults had audiometric notches, suggesting a high prevalence of noise-induced hearing loss. The prevalence of notches was higher among males. Almost one in four U.S. adults who reported excellent or good hearing had audiometric notches. Among participants who reported exposure to loud noise at work, almost one third had a notch. Noise-induced hearing loss is a significant, often unrecognized public health problem in the United States.

8:55

2aNS2. Hearing health across the lifespan: Prevention begins in early in life. Deanna K. Meinke (Audiol. & Speech-Lang. Sci., Univ. of Northern Colorado, 501 20th St., Gunter Hall 1500, Campus Box 140, Greeley, CO 80639, Deanna.Meinke@unco.edu)

Hazardous noise exposures begin early in life and evidence of noise-induced hearing loss (NIHL) and tinnitus begins to appear in adolescence and early adulthood. In animal studies, these early noise exposures can cause permanent loss of greater than 50% of cochlear-nerve/hair-cell synapses that do not recover on their own and may contribute to greater age-related hearing loss in the future. All of the landmark studies over the past 75 years that identified the risk of NIHL in children and adolescents, ended with a recommendation for education or intervention to address the problem. In 1990, a consensus statement on Noise and Hearing Loss from the National Institutes for Health called for "a comprehensive program of education regarding the causes and prevention of NIHL should be developed and disseminated, with specific attention directed toward educating school-age children." Today, there continues to be a need to nationally coordinate public health efforts to systematically address the prevention of NIHL in youth using evidence-based hearing health promotion based upon health communication science. Dangerous Decibels® is a successful evidence-based intervention program that has been demonstrated to be effective in changing knowledge, attitudes, beliefs and behaviors of both youth and adults for the prevention of NIHL and tinnitus.

9:15

2aNS3. Hearing health for adults. William J. Murphy (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov)

Recent analyses of the National Health and Nutrition Examination Survey estimate that 14% of adults between ages 20 and 69 have hearing loss defined as average hearing thresholds worse than 25 dB HL across four frequencies 0.5, 1, 2, and 4 kHz. Noise exposure is a primary cause of preventable hearing loss in the United States; an estimated 22 million American workers are exposed to hazardous noise every year. Given its associated implications on communication, education, employment opportunities, job performance, injury-risk, depression, and anxiety—hearing loss places a significant burden on society. Preventive initiatives, workplace hearing conservation programs, and habilitation/rehabilitation efforts need to be tailored to specific populations, stages of life, and hearing risk factors. Innovative hearing health programs must be developed, evaluated, and implemented. This presentation will discuss innovations in hearing loss prevention and promote hearing health for adults in the occupational and non-occupational settings.

9:35

2aNS4. Development and implementation of a military auditory fitness-for-duty test battery. Jennifer B. Tufts (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Unit 1085, Storrs, CT 06269, jennifer.tufts@uconn.edu), William Audette, Jed Wilbur, and Odile Clavier (Creare LLC, Hanover, NH)

Hearing loss is prevalent throughout the active duty military forces. However, there are no validated standards for assessing a service member's functional hearing ability and how it relates to his or her ability to accomplish the mission. In many cases, hearing-critical tasks take place in non-optimal environments with background noise or competing speech from multiple sources. Presently, hearing abilities are primarily assessed via pure-tone audiometry, without regard for the hearing-critical needs and noise environment of the individual. The development of a test battery to evaluate functional hearing ability is a highly sought-after goal. With support from the Department of Defense, Creare LLC and the University of Connecticut are developing the MILSINT, a new sound-recognition-in-noise test geared specifically for active duty personnel. The MILSINT is one component of an auditory fitness-for-duty test battery that at present also includes the Military HINT, a version of the Hearing In Noise Test (HINT) that uses military-specific phrases. The MILSINT and Military HINT are administered using a tablet interface and wireless sound-attenuating headset developed by Creare, LLC., making these tests highly portable and versatile. This presentation will describe the development of the test battery, its physical implementation, and performance data gathered to date.

9:55–10:10 Break

10:10

2aNS5. Progress in assessing and limiting occupational noise exposures in the United States. Richard Neitzel, Benjamin Roberts, Stephanie Sayler (Dept. of Environ. Health Sci., Univ. of Michigan, 1415 Washington Heights, 6611 SPH I, Ann Arbor, MI 48103, rneitzel@umich.edu), Wenting Cheng, and Bhramar Mukherjee (Dept. of Biostatistics, Univ. of Michigan, Ann Arbor, MI)

Occupational noise exposures in the United States (US) have resulted in a substantial occupational health burden on US workers. Despite the presence of regulations mandating exposure assessment and hearing loss prevention measures for highly-exposed workers, the prevalence of noise-induced hearing loss continues to be high. However, several recent efforts may result in progress in reducing exposures to, and health impacts from, occupational noise. First, the American Conference of Governmental Industrial Hygienists has recently proposed revisions to the organization's Threshold Limit Value (TLV) for Audible Sound, formerly referred to as noise. The proposed revised TLV includes updated documentation as well as notes related to non-auditory health effects of noise (including cardiovascular disease and injuries), which have not previously been considered in an occupational noise exposure limit. Second, a measurement-based national Job Exposure Matrix for Occupational Noise in the US and Canada has been developed at the University of Michigan and is now publicly available (noisejem.sph.umich.edu). This resource represents a useful tool for better understanding exposures to noise by job and industry. These two efforts, along with others, represent important progress in enhancing occupational health by protecting workers from the adverse health effects of noise.

10:30

2aNS6. Have the NIOSH age correction tables gone stale? Gregory Flamme (SASRAC, 2264 Heather Way, Forest Grove, OR 97116, gflamme@sasrac.com), Kristy Deiters (SASRAC, Portage, MI), William J. Murphy, Christa Themann (National Inst. for Occupational Safety and Health, Cincinnati, OH), and Mark R. Stephenson (SASRAC, Loveland, OH)

Occupational Hearing Conservation programs permit subtracting cross-sectional trends in hearing sensitivity from the changes observed with an exposed worker. Regulatory agencies in the U.S. define the expected cross-sectional trend using the NIOSH age correction tables, which summarized mean differences in comparatively small groups of men and women 50 years ago. At all ages, hearing sensitivity is better now than it was 50 years ago, and important demographic characteristics (e.g., race/ethnicity) that predict cross-sectional trends were not included in the NIOSH tables. Quantile regression results from the National Health and Nutrition Examination Survey (NHANES) were used to derive revised age correction tables that can be applied to men or women across a wider range of ages and race/ethnicity categories. These age correction tables and comparisons with the prior NIOSH tables will be presented.

10:50

2aNS7. Individual differences in suprathreshold hearing and relationship to cochlear synaptopathy. Hari M. Bharadwaj (Speech, Lang., & Hearing Sci., and Biomedical Eng., Purdue Univ., 715 Clinic Dr., Lyles-Porter Hall, West Lafayette, IN 47907, hbharadwaj@purdue.edu)

Threshold audiometry, although the foundation of current clinical hearing evaluations, provides limited information about suprathreshold hearing. Over the last few years, we have documented that even among listeners with normal thresholds and no hearing complaints, large individual differences exist in the ability to perceive subtle temporal features of clearly audible sounds, and to selectively process target speech in the presence of competing sounds. Furthermore, we find that these suprathreshold perceptual differences correlate with suprathreshold physiological measures from the brainstem and auditory nerve suggesting that perceptually-relevant differences may be present early along the auditory pathway. As a candidate mechanism explaining these observations, animal studies of acoustic overexposure and aging have robustly demonstrated that the afferent synapses and nerve terminals innervating the cochlea are especially vulnerable to damage. Interestingly, even a significant loss of cochlear synapses ("synaptopathy") does not lead to changes in audiometric thresholds. However, synaptopathy, if present in humans may contribute to differences in suprathreshold hearing especially in noisy environments. This presentation will summarize human evidence that is indirectly in support of the notion, and describe our ongoing efforts to test this hypothesis further and translate markers for synaptopathy from small laboratory animals to humans.

Contributed Papers

11:10

2aNS8. The non-auditory health effects of chronic noise exposure: A review. Jennifer Scinto (SoundSense, LLC, PO Box 1360, Wainscott, NY 11975, jennifer@soundsense.com), Bonnie Schnitta (SoundSense, LLC, East Hampton, NY), John Durant, and Neelakshi Hudda (Civil and Environ. Eng., Tufts Univ., Medford, MA)

This paper serves as a review of the most recent literature on the effects of chronic noise exposure on the health of human listeners. The auditory effects of occupational and other exposure to noise have been well researched and the prevention and treatment of noise-induced hearing loss is largely understood. However, as the populations of urban areas and large city centers grow, the interest in the non-auditory health effects of exposure to environmental noise such as transportation noise, aircraft noise, and other community noise has also increased. Within the past few years, several new studies have been conducted strengthening the evidence for a link between exposure to high levels of environmental noise and ill-health, especially with regards to cardio-vascular and endocrine health, immune function, sleep loss, and mental health. The research covered in this review is inclusive of human experimental, epidemiological, and mechanistic studies. Additional research will be needed to completely assess and understand the increasingly present threat to human health that is chronic environmental noise. This review will be used as the foundation for a future study examining the correlation between ill-health and noise pollution from a nearby airport.

11:25

2aNS9. Psychophysiological responses during cognitively demanding work in subjectively annoying background noise. Jordan N. Oliver, Weonchan Sung, Patricia Davies, and Alexander L. Francis (Purdue Univ., Speech, Lang. & Hearing Sci., Purdue University, West Lafayette, IN 47907, oliver49@purdue.edu)

People who work in noisy environments are at greater risk for stress-related diseases, including hypertension and stroke, even when noise levels are too low to damage hearing. Such noise may be harmful, especially to noise-sensitive individuals, because the psychological annoyance that it causes induces physiological stress responses that are damaging to health over the long term. This study was designed to investigate the link between subjective noise annoyance and physiological measures of arousal and displeasure due to the presence of background noise. Cardiovascular, electrodermal, respiratory, and facial muscular activity were recorded from 32 listeners during the completion of a demanding working memory task under different listening conditions. Participants completed four levels of memory task demand in silence and in two different continuous noises similar to that produced by HVAC equipment. Both noises were comparable in terms of loudness and presentation level (54–60 dBA) but differed in perceived annoyance (based on ratings from a panel of listeners in a previous study) and in acoustic properties associated with noise annoyance (roughness, tonality, and sharpness). Behavioral measures of memory task performance, noise sensitivity, personality traits, and subjective effort will be presented and related to physiological measures, and implications for future research will be discussed.

TUESDAY MORNING, 8 MAY 2018 GREENWAY J, 9:40 A.M. TO 11:25 A.M.

Session 2aPA

Physical Acoustics: Infrasound for Global Security I

Philip Blom, Chair
Los Alamos National Laboratory, Los Alamos National Laboratory, PO Box 1663, Los Alamos, NM 87545

Chair's Introduction—9:40

Invited Papers

9:45

2aPA1. Reflections on Los Alamos infrasound research. Rodney W. Whitaker (Earth and Environ. Sci., Los Alamos National Lab., EES 17 MS F665, LANL, Los Alamos, NM 87545, rww@lanl.gov)

Researchers at Los Alamos National Laboratory began work in infrasound research in 1982 under Department of Energy funding. This beginning was to investigate the generation and propagation of surface ground motion generated atmospheric infrasound signals. Infrasound arrays were operated to measure the infrasound signals at near regional distances. The program was successful and continued until the end of US underground nuclear testing in 1992. During this time, data were collected on earthquakes, bolides and on various man-made high explosive tests. As work on the Comprehensive Nuclear Test Ban Treaty was continuing, Los Alamos, working with Sandia National Laboratory, fielded a prototype infrasound array following Comprehensive Nuclear Test Ban Treaty guidelines, which operated from mid 1997 to 2007. More recent work emphasizes seismo-acoustic research in explosion monitoring, improving infrasound propagation models for better source location and analysis and the development of Python based tools. In this talk, I will highlight some aspects of the earlier and current Los Alamos activity.

10:05

2aPA2. Progresses with the International Data Centre Infrasound System. Pierrick Mialle (CTBTO, Vienna Int. Ctr., P.O. Box 1200, Vienna 1400, Austria, pierrick.mialle@ctbto.org) and Nimar Arora (Bayesian Logic, Inc., Union City, CA)

The International Data Centre (IDC) advances its methods and continuously improves its automatic system for infrasound technology. The IDC focuses on enhancing the automatic system for the identification of valid signals and the optimization of the network detection threshold by identifying ways to refine signal characterization methodology and association criteria. An objective of this study is to reduce the number of associated infrasound arrivals that are rejected from the automatic bulletins when generating the reviewed event bulletins (REB). Progresses related to several ongoing projects at the IDC will be review:—improving the detection accuracy at the station processing stage by introducing the infrasound signal detection and interactive review software DTK-(G)PMCC (Progressive Multi-Channel Correlation) and by evaluating the performances of detection software;—development of the new generation of automatic waveform network processing software NET-VISA to pursue a lower ratio of false alarms over GA (Global Association) and a path for revisiting the historical Infrasound Reference Event Database (IRED). The IDC identified a number of areas for future improvement of its infrasound system that will be addressed here.

10:25

2aPA3. What does infrasound signal duration tell us about signal propagation distance and waveguide structure? David Green and Alexandra Nippress (AWE Blacknest, Brimpton Common, Reading RG7 4RS, United Kingdom, dgreen@blacknest.gov.uk)

A global network of microbarograph arrays is currently being constructed to support verification of the Comprehensive Nuclear-Test-Ban Treaty. The identification of explosively generated infrasound signals that have propagated thousands of kilometers to the arrays, and the subsequent association of signals across the sparse network, remains a challenge. One signal parameter that has not been extensively studied, but may assist in source identification procedures, is the signal duration. The durations of 42 high signal-to-noise ratio signals from 35 near-surface explosions, recorded at distances of between 25 and 6300 km, exhibit a weak relationship with source-to-receiver range; longer propagation paths result in longer signal durations. The variation in signal duration at a given source-to-receiver range depends upon the atmospheric waveguide structure. At propagation distances greater than 2000 km, long duration signals are generated within weak waveguides characterized by a small excess in stratospheric effective sound speed compared to that at the ground surface. These waveguides permit signal propagation with high celerity (up to 335 m/s). Shorter duration signals at these distances occur in stronger waveguides that exhibit greater along-path variability; these signals do not exhibit celerities greater than 315 m/s. The utility of signal duration within the context of explosion monitoring is explored.

10:45

2aPA4. Exploiting both output channels of seismically decoupled infrasound sensors for enhanced transient detection and direction-of-arrival estimation. William G. Frazier (Hyperion Technol. Group, Inc., 3248 West Jackson St., Tupelo, MS 38801, gfrazier@hyperiontg.com)

Some infrasound sensors have internal sensing capabilities that remove or permit subsequent removal of much of the effect of ground motion on the measured pressure fluctuations. This presentation explores possible enhancements to transient acoustic signal detection and direction-of-arrival estimation when data corresponding to the ground motion and pressure fluctuation signals are analyzed simultaneously using statistical array signal processing techniques. Data from three 4-element infrasound sensor arrays that were deployed to detect explosive events occurring from 10 km to 20 km distant are used as the basis for the study.

11:05

2aPA5. The application of acoustic particle velocity sensors to airborne infrasonic signals. W. C. Kirkpatrick Alberts (U.S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20723, william.c.alberts4.civ@mail.mil)

Infrasound arrays, because of the wavelengths involved and the necessity for minimal spatial aliasing, can be quite large, e.g. an array layout often used for beamforming for frequencies up to 10 Hz is a centered, 17-m radius equilateral triangle. Available real estate is often limited such that optimally sized arrays are not possible. An acoustic particle velocity sensor, in principle, can perform the same job as a large infrasound array, but in a much smaller package (less than the size of a single infrasound sensor) since it provides, in post processing, a direct measurement of the wavefront normal. However, localized meteorological perturbations to the wavefront normal could cause significant bearing estimate errors when using a particle velocity sensor. Recently, as part of the characterization of a mobile infrasound source and during the collection of sounds from space station resupply rockets, acoustic particle velocity sensors were positioned next to infrasound sensors in order to test the concept of using these sensors for accurate pointing at infrasound frequencies. Results from these experiments will be discussed.

Session 2aPP

Psychological and Physiological Acoustics, Speech Communication, and Signal Processing in Acoustics: Phase Locking and Rate Limits in Electric Hearing

Mathias Dietz, Chair
National Centre for Audiology, Western University, 1201 Western Road, London, ON N6G 1H1, Canada

Chair's Introduction—8:00

Contributed Paper

8:05

2aPP1. The effects of sound coder carrier rate and modulation bandwidth on voice pitch perception in cochlear implant users. Damir Kovacic (Depertment of Phys., Univ. of Split, Faculty of Sci., R.Boskovica 33, Split 21000, Croatia, Damir.Kovacic@pmfst.hr) and Chris James (Cochlear SAS France, Toulouse, France)

We employed the dual filter-bank "STEP" coder to separately control the spectral and temporal modulation resolution of analysis channels. Previously we compared vowel pitch ranking and gender classification with eight subjects using enhanced modulation at F0—including across-channel synchronised modulation to the ACE coder. There was no significant improvement using modulation enhanced coding versus ACE across subjects. In a follow-up experiment we looked at the effect of stimulation rate on voice pitch perception. Since there are large inter-subject differences in overall temporal pitch acuity we hypothesised that some subjects' performance may be more greatly influenced by carrier rate than others, or that some subjects may find sound quality satisfactory with lower carrier rates than those in their clinical processors. We used a version of STEP with a very short temporal envelope analysis window of 2 ms which allows a very low latency real-time processing implementation and large maximum modulation bandwidth. Subjects were tested using carrier rates of 1000, 500 and 250 pps/ch with modulation bandwidths controlled via low-pass filtering. Pilot data indicated that the new low-latency coder provides very good sound quality compared to ACE using 1000 pps/ch or 500 pps/ch. Also the modulation bandwidth could be tuned at different carrier rates to optimize voice pitch perception based on temporal cues. This opens the potential for lower stimulation rates to be used in CI coding while maintaining optimal temporal resolution.

Invited Papers

8:20

2aPP2. Phase-locking in acoustic hearing: Upper limits, enhancement, and transformations. Philip X. Joris (NeuroSci., Univ. of Leuven, Lab Auditory Neurophysiol, Herestraat 49 box 1021, Leuven B-3000, Belgium, philip.joris@med.kuleuven.be)

Phase-locking is a remarkable neural property of various mechanoreceptor-based sensory systems, and the neural coding of cochlear vibrations is the primary model system to study this phenomenon. Transduction and synaptic transmission by cochlear inner hair cells enables but also limits temporal coding in the auditory nerve. I will review phase-locking to sound fine-structure in the auditory nerve of various species, including humans, assessed with different types of sound stimuli and using different metrics. The upper-frequency limit of phase-locking in cell populations in the central nervous system is lower than that in the auditory nerve, and at the same time, the nature of the temporal code is transformed in these neurons: phase-locking is enhanced in that it is more consistent and has decreased jitter, and it has profound effects on the average discharge rate of various populations of neurons, both monaurally and binaurally. Similar phenomena (changing upper limits, enhancement, and rate effects) are observed in the temporal coding of envelopes and click trains. While it is increasingly clear that many cell groups in the auditory brainstem have specialized mechanisms towards temporal coding, the relationship of the resulting response properties to perception remains unclear.

8:40

2aPP3. Comparison of temporal processing in the auditory brainstem neurons between acoustic and electrical hearing. Michaela Müller, Barbara Beiderbeck, Benedikt Grothe, and Michael Pecka (Biology II; Neurobiology, Ludwig-Maximilians Univ. Munich, Grosshaderner Strasse 2, Martinsried 82152, Germany, pecka@bio.lmu.de)

Spatial hearing is essential for communication as well as navigation in everyday life. Unfortunately, sound localization in these complex environments is severely limited in patients with bilateral cochlear implants (CIs) and its restoration thus remains one of the central obstacles of CI research. Spatial sensitivity is generated by neurons in the brainstem that detect differences in the arrival time between excitatory and inhibitory inputs from the two ears on the scale of only microseconds. However, the mechanisms underlying this precise temporal integration of individual inputs and potential differences in temporal precision during electrical stimulation are still not understood. Here, we obtained *in vivo* electrophysiological recordings from single neurons in the brainstem of Mongolian Gerbils

(*Merionesunguiculatus*). We characterized the temporal precision of action potential firing in response to click trains in binaural neurons of the Lateral Superior Olive and their upstream inputs (auditory nerve, cochlear nucleus and medial nucleus of the trapezoid body). To assess potential differences in temporal precision between acoustic and electrical stimulation, complementary data are obtained in normal hearing animals and animals that underwent cochlear implantation. Ultimately, we aim to identify basic principles of binaural integration in the auditory brainstem during acoustic and CI-based spatial hearing.

9:00

2aPP4. Measurements and models of auditory nerve spike rate and timing to electrical stimulation. Ian C. Bruce (Dept. of Elec. & Comput. Eng., McMaster Univ., Rm. ITB-A213, 1280 Main St. W, Hamilton, ON L8S 4K1, Canada, ibruce@ieee.org)

Physiological measurements of the response of auditory nerve (AN) fibers to electrical stimulation from a cochlear implant (CI) have in general exhibited much higher maximal entrainment rates and higher temporal precision than is observed for acoustical stimulation. However, it appears that this superior rate coding in the AN for electrical stimulation does not translate to better coding of rate in the midbrain or for the percept of temporal pitch. Similarly, the enhanced temporal precision for electrical stimulation does not lead to improved coding of interaural timing differences. In this talk, I will review physiological data and computational model simulations that indicate that the superior spike rate and timing representation for CI stimulation may only hold when considering the response of *some* single AN fibers to *repeated identical stimuli*. In fact, there is a large heterogeneity in spiking responses that can change dramatically as a function of stimulus current level. A multi-compartmental computational model will be used to demonstrate how the electrode- fiber geometry and the membrane biophysics could be contributing to these phenomena. I will also discuss the implications of the likely population response of the AN to CI stimulation for brainstem processing. [Work supported by NSERC Discovery Grant 261736.]

9:20

2aPP5. Effect of site of spike generation on temporal coding in the electrically stimulated auditory nerve. Suyash N. Joshi (Hearing Systems Group, Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Bygning 352, Kongens Lyngby 2800, Denmark, sjoshi@elektro.dtu.dk)

At least two sites of spike generation are observed in the electrically stimulated auditory nerve, namely, the peripheral axon and the central axon. The peripheral axon responds to the cathodic charge and the central axon to the anodic charge with an approximate spike-latency difference of 200 μs. The peripheral axon also shows longer refractory periods than the central axon. In this study, a phenomenological model of the electrically stimulated auditory nerve described in Joshi *et al.* [JARO 18(2), pp. 323–342] was used to understand the effect of spike generation sites on the temporal coding. Modulation detection thresholds for different carrier pulse rates were predicted from simulated responses of the auditory nerve, and refractoriness of both the axons was varied to test its effect on temporal coding. The results show that an increase in pulse rate increased the number of spikes generated at the central axon, hence decreasing the modulation detection thresholds. Refractoriness also played a crucial role in encoding the stimulus envelope such that extended refractory periods resulted in enhanced envelope coding. Increase in refractoriness at the central axon improved the envelope encoding. Implications of these findings for developing new stimulation strategies will be discussed.

9:40–9:55 Break

9:55

2aPP6. Modeling Cochlear Implant processing and bilateral performance in acoustic environments with multiple sources. H. Steven Colburn (Hearing Res. Ctr.; Biomed. Eng., Boston Univ., 44 Cummington Mall, Boston, MA 02215, colburn@bu.edu)

The work reported here focuses on the use of cochlear implants in situations with multiple sources, especially speech sources, and in other difficult environments. After a review of previous measurements and models of cochlear implant (CI) stimulation, recent modeling studies of speech processing in conditions with multiple inputs in complex acoustic environments will be discussed. As noted above, complex environments considered include conditions with multiple speech sources, and this talk focuses on effects of spatial separation and voice pitch differences in confusions related to source separation. These effects include both energetic and informational masking. Multiple implant factors will be addressed, including interaural variability in the timing of the left and right stimuli, the effects of independent left and right automatic gain control, and the interactions with and basic role of pulse rate in the implants. [Work supported by NIDCD 5 R01 DC000100.]

10:15

2aPP7. Comparisons of electric and acoustic ITD coding in normal hearing gerbils. Maike Vollmer (Otolaryngol., Otto von Guericke Univ. Med. School, Leipziger Strasse 44, Magdeburg 39120, Germany, maike.vollmer@med.ovgu.de)

Small differences in the arrival time of sound between the two ears [interaural time differences (ITDs)] provide important cues for directional hearing and speech understanding in noise. Deaf subjects with bilateral cochlear implants (CIs) show relatively poor sensitivity to ITDs. It is unclear whether these limitations are due to differences in binaural brain circuits activated by electric compared to acoustic stimulation, mismatches in electric activation site across ears, or deafness-induced degradations in neural ITD processing. To identify potential limitations of electric ITD coding, we compared electric and acoustic ITD coding (i.e., ITD tuning and discrimination thresholds) in the same population of auditory brainstem and midbrain neurons in normal hearing gerbils. When compared in the same neurons, ITD coding to acoustic stimulation did not predict coding to electric stimulation. However, on a population level, neurons demonstrated surprising similarities in acoustic and electric ITD processing. Importantly, even short periods of deafness (2 weeks) severely degraded electric ITD processing. The findings suggest that discrepancies in ITD discrimination between bilateral CI users and normal hearing listeners are primarily due to deafness-induced changes in neural ITD processing rather than differences in the binaural brain circuits activated by either electric or acoustic stimulation.

10:35

2aPP8. Towards increasing timing sensitivity in electric hearing. Bernhard Laback (Acoust. Res. Institute, Austrian Acad. of Sci., Wohllebengasse 12-14, Vienna 1040, Austria, bernhard.laback@oeaw.ac.at), Sridhar Srinivasan (Acoust. Res. Institute, Austrian Acad. of Sci., Wien, Wien, Austria), Martin Lindenbeck, Maike Ferber (Acoust. Res. Institute, Austrian Acad. of Sci., Vienna, Austria), and Piotr Majdak (Acoust. Res. Instiute, Austrian Acad. of Sci., Wien, Austria)

Sensitivity of bilateral cochlear-implant (CI) listeners to interaural time differences (ITDs) in electric pulse trains is degraded compared to normal-hearing (NH) listeners presented with ITDs in pure tones. This degradation manifests both as an elevated ITD threshold and upper perceptual limit of stimulation rates. Similar limitations were observed for temporal pitch, despite the difficulty to disentangle temporal and place pitch cues in NH listeners. We tested the hypothesis that ITD and monaural rate-pitch sensitivity of CI listeners at high rates of electric stimulation can be improved by introducing extra pulses with short inter-pulse intervals (SIPIs) at low rates in amplitude modulated high-rate pulse trains. Results show that SIPIs significantly improved ITD and monaural rate-pitch sensitivity at low modulation depths. These similar improvements suggest a possible overlapping benefit of SIPIs for CI listeners in everyday environments requiring high rates for encoding speech. Considering the documented effects of neural deprivation, its potential reversal by training, and the potential for NH-like timing sensitivity as observed in exceptional CI listeners, we began to investigate the effects of vision-induced ITD training in CI listeners. Preliminary results will be discussed in the light of the timing sensitivity limitations in electric hearing.

10:55

2aPP9. Restoring binaural and spatial hearing in cochlear implant users. Ruth Litovsky, Alan Kan, Tanvi Thakkar, Sean R. Anderson (Waisman Ctr., Univ. of Wisconsin - Madison, 1500 Highland Ave., Waisman Ctr. Rm. 521, Madison, WI 53705, litovsky@waisman.wisc.edu), Z. Ellen Peng (Waisman Ctr., Univ. of Wisconsin - Madison, Wisconsin, U.S. Minor Outlying Islands), and Thibaud Leclère (Waisman Ctr., Univ. of Wisconsin - Madison, Madison, WI)

Adults and children who receive bilateral cochlear implants (BiCIs) have the potential to benefit from the integration of inputs arriving at the brain from both ears. Several factors play a key role in determining if patients will demonstrate binaural sensitivity. We are exploring these factors using two experimental approaches. In the first approach, BiCI users receive pulsatile stimulation to specific pairs of electrodes using research processors that synchronize stimulation with fidelity. We vary stimulus parameters such as temporal fine structure and envelope cues, places of stimulation along the cochleae, and number of electrodes to find parameters that maximize sensitivity to interaural differences for each patient. We also investigate the role of the electrode-neuron interface which is affected by numerous factors including neural health. In a second stimulation approach, we use clinical speech processor to deliver binaural stimulation designed specifically for that patient based on their clinical MAP. In these studies, we are using both standard psychophysics and eye gaze paradigms to understand the underlying processing involved in binaural and spatial hearing. This combined approach is enabling us to design multi-channel multi-rate stimulation strategies aimed at restoring binaural sensitivity and preserving speech understanding.

11:15

2aPP10. Temporal constraints on neural and psychophysical sensitivity to interaural level difference cues to sound source location. Andrew D. Brown and Daniel J. Tollin (Dept. of Physiol. & Biophys., Univ. of Colorado School of Medicine, University of Colorado Anschutz Medical Campus, 12631 E 17th Ave., Aurora, CO 80045, Daniel.Tollin@ucdenver.edu)

Interaural differences in the timing (ITD) and level (ILD) of impinging sounds carry critical information about source location. However, in everyday listening environments, sounds are often decorrelated between the ears by reverberation and background noise, degrading the fidelity of both ITD and ILD cues. Similar distortions to ITD and ILD are also experienced by hearing-impaired humans who use hearing aids or cochlear implants, as these devices also degrade temporal and intensive features of the signal at each ear. Here, we demonstrate that behavioral ILD sensitivity (in humans) and neural ILD sensitivity (in single neurons of the chinchilla auditory midbrain) remain robust under stimulus conditions that render ITD cues undetectable. Neural and behavioral data were compared to the outputs of a model of ILD processing with a single free parameter, the duration of excitatory-inhibitory interaction. Behavioral, neural, and modeling data collectively suggest that ILD sensitivity depends on binaural integration of excitation and inhibition within a $\gtrsim$3-ms temporal window, significantly longer than observed in lower brainstem neurons. This relatively slow integration potentiates a unique role for the ILD system in spatial hearing that may be of particular importance when informative ITD cues are unavailable. [Work supported by F32-DC013927 [ADB] and R01-DC011555 [DJT].]

11:35–11:55 Panel Discussion

Session 2aSC

Speech Communication, Animal Bioacoustics, and Psychological and Physiological Acoustics: Adapting Methods and Models for Vocal Production Across Human and Non-Human Primate Species

Benjamin Munson, Cochair
University of Minnesota, 115 Shevlin Hall, Minneapolis, MN

Michael B. Wilson, Cochair
Acoustics, Penn State, 1649 Highlandon Ct, State College, PA 16801

Mary E. Beckman, Cochair
Department of Linguistics, Ohio State University, 1712 Neil Ave., Columbus, OH 43210

Chair's Introduction—8:40

Invited Papers

8:45

2aSC1. Tracking chimpanzee pant-hoot changes across time and space. Michael Wilson (Anthropology, Univ. of Minnesota, 395 Humphrey Ctr., 301 19th Ave. S, Minneapolis, MN 55455, wilso198@umn.edu), Lisa R. O'Bryan (Dept. of Ecology, Evolution and Behavior, Univ. of Minneapolis, Sugar Land, Texas), Andrew R. Plummer (Comput. Sci. and Eng., Ohio State Univ., Columbus, OH), Mary E. Beckman (Linguist, Ohio State Univ., Columbus, OH), and Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minneapolis, Minneapolis, MN)

Humans differ strikingly from other primates in the capacity for vocal learning. How and why such vocal flexibility evolved remains puzzling. Evidence of geographic variation in the pant-hoot calls of chimpanzees (*Pan troglodytes*) suggests that chimpanzees have some capacity for vocal learning, which is intriguing given the close phylogenetic relationship between humans and chimpanzees. Many questions remain, however, about the various factors that might contribute to variation in acoustic structure of pant-hoot calls, including body size, health, genetic relatedness, and within-individual variation. We are currently examining these factors in a study of longitudinal recordings from individuals in two neighboring chimpanzee communities in Gombe National Park, Tanzania. As in studies of sound change in human social groups, we need to understand the articulatory mechanisms that produce the vocalizations, to interpret acoustic variation against the backdrop of factors including body size, sex, and age. The same considerations in studies aimed at understanding the ontogeny of human vocalizations have prompted the development of age-specific articulatory synthesis models. We are adapting such models to design analysis-by-synthesis methods for our cross-population longitudinal study, and have organized this session to explore analogous methods for studying other vocalization types in chimpanzees and other non-human primate species.

9:05

2aSC2. Supervised and unsupervised machine learning approaches to classifying chimpanzee vocalizations. Nisarg P. Desai (Anthropology, Univ. of Minnesota, 395 Humphrey Ctr., 301 19th Ave. S, Minneapolis, MN 55455, desai054@umn.edu), Clarence Lehman (Ecology, Evolution, and Behavior, Univ. of Minnesota, Saint Paul, MN), Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), and Michael Wilson (Anthropology, Univ. of Minnesota, Minneapolis, MN)

Quantitative tools for classifying vocal repertoires have been constantly evolving with developments in machine learning and speech recognition research as well as increasing computing power. There are two main methodological considerations in classifying vocalizations: (i) choosing the classification technique and (ii) choosing the features for classification. Current state-of-the-art classification techniques are artificial neural networks (ANNs), support vector machines (SVMs), and ensemble methods like random forests (RFs). Current state-of-the-art features from speech recognition research include mel frequency cepstral coefficients (MFCCs). Bioacoustics researchers have applied these tools to problems including individual-, species-, and call-type identification, and vocal repertoire classification. However, researchers studying non-human primate vocalizations have only recently started adopting these approaches and none have applied them to study chimpanzee vocalizations. Here, we analyze vocalizations recorded in Gombe National Park, Tanzania. First, we use supervised classification techniques (ANNs, SVMs, and RFs) that involve training the models based on predefined call-types to evaluate the classification accuracy. Second, we use unsupervised techniques (that do not require prior knowledge of call-types), namely, K-means clustering, and self-organizing neural networks to identify discrete call types. We discuss the results from both supervised and unsupervised techniques and their strengths over traditional methods.

9:25

2aSC3. Acoustic communication by vocal tract modulation. Brad H. Story (Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., P.O. Box 210071, Tucson, AZ 85721, bstory@email.arizona.edu)

In both human and nonhuman animals, the airway system may serve as an instrument for acoustic communication. Flow-induced tissue vibrations and noise sources generate the acoustic excitation, whereas the configuration of the vocal tract provides a variable resonant filter system that transforms the excitation into a "message." This presentation will describe the development of a vocal tract model in which the effects of articulatory movements that produce speech are generated by specifying independent acoustic events along a time axis. These events consist of directional changes in the first three resonance frequencies of an acoustically-neutral airway configuration and are transformed, via acoustic sensitivity functions, into time-varying modulations of the vocal tract shape. The duration of each event may be considerably overlapped in time with other events to produce efficient transmission of information through the effects of coarticulation. The model will be used to demonstrate construction of syllables, words, and phrases based on a range of underlying idiosyncratic "neutral'' vocal tract configurations representative of typical and unusual airway systems. [Research supported by NIH R01-DC011275, NSF BCS-1145011, and NIH R01-DC006282.]

9:45

2aSC4. Articulatory modeling of human and non-human vocal production. Kiyoshi Honda and Ju Zhang (School of Comput. Sci. and Technol., Tianjin Univ., 135, Yaguan Rd., Jinnan Dist., Tianjin 300350, China, khonda@sannet.ne.jp)

This paper reviews two of the many questions that might be addressed by adapting articulatory models of vocal production across primate species. The first concerns vocal-fold shaping at phonation. The human folds have been modeled to have uniform cross-sectional shapes with round edges along the longitudinal axis, whereas MRI slices reveal an obvious vertical prominence of the mucosa near each vocal process. This shape, due to depressed vocal processes in adduction, somewhat resembles the pointed edges of the vocal membranes in nonhuman primates, which are incorporated in some models of chaotic phonation in those species. This suggests a model of vocal-fold vibration propagating back to the posterior glottis, or coupled oscillation for irregularities in human voices. The second concerns tongue-lip coordination. In human vowel inventories, tongue retraction typically accompanies rounding. In a case report, a trained Chimpanzee was able to produce back vowels /a, o, u/ in whispering, but could not coordinate these articulatory positions with voiced phonation. This suggests an innate neural circuit for patterned tongue-lip coordination as seen in sucking, and such a basic neural unit may be active in humans to simplify the articulatory control of back rounded vowels. [Work supported by NSFC No. 61573254.]

10:05 – 10:20

10:20

2aSC5. The challenges of developing articulatory synthesis models of early vocal production in humans. Andrew R. Plummer (Ohio State Univ., 2015 Neil Ave., Columbus, OH 43210, plummer.321@osu.edu)

At birth, human infants can cry, but they cannot articulate anything like the sound patterns of their mothers' speech that they have been hearing in the last month or so in the womb. Learning to speak involves a re-tuning of the muscle systems for breathing, crying, and suckling in order to be able to articulate a rapid sequence of exquisitely coordinated movements of the speech articulators—gestures of the larynx, velum, tongue, and lips—in synchrony with a lengthened expiratory phase of the breathing cycle, to acoustically shape the air flowing from the lungs to produce the sound patterns that are the words of a specific language. Models of speech production developed using data on articulator movement and coordination in adults cannot be applied directly, because this learning takes place in the context of substantial changes to the size and shape of the vocal tract over the course of physical maturation. This talk describes the changes in vocal tract morphology in human development, and then reviews how articulatory synthesis systems developed for adults have been adapted to model the expanding repertoire of vocalizations that have been observed in infants with normal hearing over the first year of life.

10:40

2aSC6. Physical growth drives vocal sequence development in marmoset infants. Yisi Zhang and Asif Ghazanfar (Princeton Univ., Princeton Neurosci. Inst., Princeton, NJ 08544, yz9@princeton.edu)

The vocal sequences of marmosets exhibit many drastic changes during infancy. One of the most pronounced of these is the gradual convergence from a variety of call types in the undirected context to long distance contact calls (aka phee calls). We conjecture that such change is attributed to the interplay between the neural fluctuation and the biomechanics of the developing vocal periphery. To explore such possibility, we first quantitatively characterize vocal sequences and correlate them with animal momentary arousal levels. Based on this, we show that the diversity of respiratory and laryngeal patterns, as well as the commonly produced sequence transitions, can be generated via simple coupled oscillators with a low-dimensional input. In addition, the model predicts that as the respiratory apparatus grows, not only does the oscillatory period increase, but also the proportions of call types become biased towards phee calls. We further empirically test this hypothesis by placing the marmoset infants in a helium-oxygen (heliox) environment where the much lighter air reduces vocal effort, simulating smaller lungs. Consistent with the model prediction, the heliox manipulation reduces phee call duration and increases the proportion of other call types.

11:00

2aSC7. Chimpanzee food-associated calls display low referential potential in a wild population. Lisa R. O'Bryan (Ecology, Evolution and Behavior, Univ. of Minnesota, 13915 Lynnwood Ln, Sugar Land, TX 77498, obry0017@gmail.com), Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), and Michael Wilson (Ecology, Evolution and Behavior, Univ. of Minnesota, Minneapolis, MN)

Studies of non-human primate vocal communication commonly focus on "functionally referential signals," which are thought to function like human words, informing receivers about stimuli in the external environment. Captive studies of the food-associated "rough-grunt" of chimpanzees (*Pan troglodytes*) report that their acoustic structure varies according to food quality and even type, suggesting the existence of functionally referential communication in humans' evolutionary cousins. Nevertheless, studies of wild chimpanzees have produced mixed evidence that rough-grunts function referentially in natural contexts. The current study builds upon these findings by conducting an acoustic analysis of rough-grunts produced by wild chimpanzees at Gombe National Park, Tanzania, examining acoustic variation in call duration and peak frequency both within and between feeding bouts. We found that peak frequency, but not duration, displayed a bimodal distribution, supporting the view that rough-grunts include at least two acoustic sub-types. Nevertheless, calling bouts produced within each feeding bout reliably encompassed this full range of acoustic variation and calls did not display consistent temporal patterns of variation throughout feeding bouts. Our findings are thus inconsistent with the idea that rough-grunts label food properties. Rather, we argue that rough-grunts broadcast information about the signaler's foraging or social intentions.

11:20

2aSC8. Michael Owren's contributions to methods and models of vocal production for human and non-human primates. Drew Rendall (Univ. of NB, Sir Howard Douglas Hall 320, Fredericton, NB E3B 5A3, Canada, d.rendall@unb.ca)

This part of the special session comprises a review of some of the highlights of Michael Owren's many contributions to adapting methods and models of vocal production across primate species, punctuated by comments and discussion by the invited speakers, in preparation for the larger panel discussion at the end of the session.

11:40–12:00 Panel Discussion

TUESDAY MORNING, 8 MAY 2018 — GREENWAY H/I, 8:20 A.M. TO 10:05 A.M.

Session 2aSP

Signal Processing in Acoustics, Underwater Acoustics, Physical Acoustics, and Noise: Time Reversal Acoustics

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

Invited Papers

8:20

2aSP1. Localization of sources of noise in moving atmospheres using time reversal and geometric acoustics. 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)

Due to acoustic refraction and convection in moving, inhomogeneous atmospheres, the backazimuths of recorded signals can be significantly different from the source backazimuth, posing a significant difficulty in source localization. Although an inverse problem, localization of sources of infrasound in such atmospheres has mostly been made using the forward ray-theoretic model of sound propagation. This approach uses search grids for the unknown model parameters that are subsequently estimated by optimizing an objective function. This approach can give rise to a large computational time as well as to the issue of uniqueness, stability, and accuracy of the solutions. The current paper presents a technique which avoids these issues, and which combines time reversal and geometric acoustics for source localization in such atmospheres. This technique, which can also be useful for source localization of other forms of acoustic signals in refractive media, effectively transforms the inverse problem into a forward one by utilizing a flow reversal theorem in the ray tracing equations. Beside the atmospheric conditions, the only parameters needed in this technique are the trace velocities of the signals recorded by at least two arrays. Comparison of numerical simulations with observational data will be presented.

8:40

2aSP2. Synthetic time reversal and remote acoustic sensing of changes in a vibrating structure. David R. Dowling and Tyler J. Flynn (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109-2133, drd@umich.edu)

Synthetic time reversal (STR) is a method for blind deconvolution that is based on the time-reversal symmetry of acoustic fields. It can be used to estimate an unknown acoustic source's broadcast waveform in an unknown multipath environment from remote transducer array recordings. This presentation will review the formulation of STR, and its use in ocean acoustics and non-destructive evaluation. In particular, experimental results are presented for the remote acoustic detection of mechanical changes to a vibrating 0.3-m-square by 3-mm-thick aluminum plate in a reverberant environment. The plate has nominally clamped edges and is subject to swept-frequency base excitation from 100 to 2,000 Hz. Sound radiated from the plate with and without a mechanical change is recorded remotely with a 15-microphone linear array and processed using STR to reconstruct a single radiated-sound signal that is deconvolved from the environment's unknown reverberation response at signal-to-reverberation ratio levels of -7 to -13 dB. The corrected signals are processed to detect boundary clamping defects and various length cuts in the plate via standard statistical baseline comparison techniques. Detection results using STR are superior and more robust to geometrical uncertainties than equivalent results from conventional approaches. [Sponsored by ONR and NAVSEA through the NEEC]

9:00

2aSP3. Multiple time-reversal focusing with a virtual source array. Gihoon Byun (Korea Maritime and Ocean Univ., Korea Maritime Univ., Dongsam 2-dong, Yeongdo-gu, Busan 606-791, South Korea, gihoonbyun77@gmail.com), H. C. Song (Scripps Inst. of Oceanogr., San Diego, CA), and J. S. Kim (Korea Maritime and Ocean Univ., Busan, South Korea)

Time reversal (TR) is the process of generating a spatio-temporal focus at a probe source (PS) location by backpropagating a time-reversed version of a received signal. While TR focusing requires the PS for a coherent acoustic focus at its origin, the requirement of the PS has been partially relaxed by the concept of a virtual source array (VSA) [J. Acoust. Soc. Am. 125, 3828–3834 (2009)]. A VSA can serve as a remote platform or lens and redirect a focused field to a selected location beyond the VSA for which the field is assumed as a homogeneous medium with constant sound speed. The objective of this study is to extend VSA-based single TR focusing to simultaneous multiple focusing. This is achieved using the optimization theory by employing the multiple constraints method derived from a constraint matrix, which consists of appropriately synchronized transfer functions. It is found that simultaneous multiple focusing can be achieved with distortionless response at selected multiple locations, and its performance degrades in the presence of sound speed mismatch. Possible applications for the VSA-based TR focusing are discussed, and numerical simulation results are presented

Contributed Papers

9:20

2aSP4. Time reversed elastic nonlinearity diagnostic applied to stress corrosion cracking in stainless steel. Sarah M. Young, Brian E. Anderson, and Stephen M. Hogg (Phys. and Astronomy, Brigham Young Univ., ESC N203, Provo, UT 84601, sarahmyoung24@gmail.com)

The Time Reversed Elastic Nonlinearity Diagnostic (TREND) is an inspection technique that enables one to locate damage in a structure using time reversal processing. TREND has been used to detect various types of damage in metals and rocks, but little has been done specific to stress corrosion cracking (SCC) other than the work of Anderson *et al.* J. Acoust. Soc. Am. **141**, EL76–EL81 (2017). SCC in stainless steel, especially around welds, is produced as a result of elevated temperature and saline conditions, and so has different properties from fatigue cracking or delaminations, both of which have been studied with TREND. In current research, welded steel samples exposed to a magnesium chloride bath were created. A number of samples were exposed for successively longer durations in order to study the corrosion damage level versus exposure time. Using TREND, samples were scanned then evaluated for indicators of local cracking at each scan point. Specific changes in spectral characteristics that are indicators of cracking were found to correlate to a longer exposure time and therefore the extent of SCC damage at that location.

9:35

2aSP5. Understanding the effect of room parameters on acoustic time reversal. Michael Denison and Brian E. Anderson (Phys., Brigham Young Univ., 485 S State St, Apt. 306, Provo, UT 84606, michael.denison23@gmail.com)

Time Reversal (TR) is a technique used to focus an acoustic signal at a particular point in space. Of the many variables that contribute to the quality of TR focusing, the most important are the number of sound sources, signal bandwidth. and properties of the medium. Although much research has been done to quantify the effects that the number of sound sources and signal bandwidth have on the TR process, little has been done in regard to the effect that room parameters have on TR. This is largely due to the difficulty involved with changing room acoustic parameters to measure their effects. We use the image source method (using the algorithm proposed by Allen and Berkley) to simulate the TR process in a variety of rooms with different acoustic and geometric room parameters. We define and calculate the maximum focal amplitude, the temporal focus quality and the spatial focus quality for each simulation. We compare the results and determine the effects of absorption and room volume on TR. We find that less absorption improves max response and spatial quality while it decreases temporal quality and that larger volumes have decreased max response and spatial quality while having increased temporal quality.

9:50

2aSP6. Using time reversal focusing for active control of magnetic resonance imaging noise. Trent Furlong and Brian E. Anderson (Dept. of Phys. & Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602, trentfurlong@gmail.com)

The noise produced by a Magnetic Resonance Imaging (MRI) scan can cause discomfort to patients. Active Noise Control (ANC) systems have been developed to help reduce the amount of noise a patient experiences. However, due to the small enclosure of the MRI machine, ANC equipment can be cumbersome to use, and is limited to non-ferrous material. Time Reversal Acoustics (TRA) is a technique that can focus sound to a selected position in space that may be far from actual sound sources. We are exploring using TRA to deliver an ANC signal by broadcasting "anti-noise" signals that are focused to the patient's ears through TRA to cancel the MRI noise. This presentation will explore the applicability of using TRA to deliver different types of ANC signals to determine limitations in this process and show some proof of concept results.

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, 7 May 2018, from 5:00 p.m. to 6:00 p.m.

The Standards Committee Plenary Group meeting will precede the meetings of the Accredited Standards Committees S1, S3, S3/SC 1, and S12, which are scheduled to take place in the following sequence:

Tuesday, 8 May 2018	11:00 a.m.–12:15 p.m.	S12, Noise
Tuesday, 8 May 2018	2:00 p.m.–3:00 p.m.	ASC S3/SC 1, Animal Bioacoustics
Tuesday, 8 May 2018	3:15 p.m.–4:30 p.m.	ASC S3, Bioacoustics
Tuesday, 8 May 2018	4:45 p.m.–5:45 p.m.	ASC S1, Acoustics

Discussion at the Standards Committee Plenary Group meeting will consist of national items relevant to all S Committees and U.S. TAGs.

The U.S. Technical Advisory Group (TAG) Chairs for the various international Technical Committees and Subcommittees under ISO and IEC, which are parallel to S1, S2, S3, and S12 are as follows:

U.S. TAG Chair/Vice Chair	TC or SC	U.S. Parallel Committee
ISO		
R. D. Hellweg, Jr., Chair P. D. Schomer, Vice Chair	**ISO/TC 43** Acoustics	ASC S1 and ASC S3
R. D. Hellweg, Jr., Chair P. D. Schomer, Vice Chair	**ISO/TC 43/SCI** Noise	ASC S12
R. W. Fischer, Chair S12	**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
M. A. Bahtiarian, Chair ASC S12	ISO/TC 43/SC 3, Underwater acoustics	ASC S1, ASC S3/SC 1 and
W. Madigosky, Chair	ISO/TC 108 Mechanical vibration, shock, and condition monitoring	ASC S2
M. L'vov, Chair	ISO/TC 108/SC2 Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles, and structures	ASC S2
D. J. Evans, Chair	ISO/TC 108/SC3 Use and calibration of vibration and shock measuring instruments	ASC S2
D. D. Reynolds, Chair	ISO/TC 108/SC4 Human exposure to mechanical vibration and shock	ASC S2 and ASC S3
D. J. Vendittis, Chair	ISO/TC 108/SC5 Condition monitoring and diagnostics of machine systems	ASC S2
IEC		
D. A. Preves and C. Walber, U.S. Technical Co-advisors	IEC/TC 29 Electroacoustics	ASC S1 and ASC S3

Meeting of Accredited Standards Committee (ASC) S12 Noise

S. J. Lind, Vice Chair ASC S12
The Trane Co., 3600 Pammel Creek Road, Bldg. 12-1, La Crosse, WI 54601 7599

D. F. Winker, Vice Chair ASC S12
ETS-Lindgren Acoustic Systems, 1301 Arrow Point Drive, Cedar Park, TX 78613

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

People interested in attending the meeting of the TAG for ISO/TC 43/SC 1, Noise, and ISO/TC 43/SC 3, Underwater acoustics, take note that meeting will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 8 May 2018.

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

TUESDAY AFTERNOON, 8 MAY 2018 NICOLLET C, 1:00 P.M. TO 4:40 P.M.

Session 2pAA

Architectural Acoustics: Interactions Between Acoustics and Architectural Design

Ana M. Jaramillo, Cochair
Ahnert Feistel Media Group, 8717 Humboldt Ave. N, Brooklyn Park, MN 55444

Adel Hinawi, Cochair
Acoustic Distinctions, One Grand Central Place, 60 East 42nd Street, Suite 2036, New York, NY 10165

Chair's Introduction—1:00

Invited Papers

1:05

2pAA1. The architect as Ally: A multi-sensory approach to design. Gregory A. Miller and Robin S. Glosemeyer Petrone (Threshold Acoust., LLC, 141 W. Jackson Boulevard, Ste. 2080, Chicago, IL 60604, gmiller@thresholdacoustics.com)

The end product of an architectural acoustic design is not the quality of the documentation or the successful achievement of a set of parameters, but rather the experience of the end users. With rare exception, the perception of acoustic characteristics cannot be separated from the users' visual and tactile experiences. To unify these sensory experiences, we have adopted an approach to acoustic design that fully embraces the architect's spatial and material vision for a building. At the same time, we seek to educate the architect so that they can embrace and internalize the acoustic goals of the building in their work. This presentation will describe our overall design approach, means of communication and demonstration used to build conviviality with our clients, and examples where acoustic and architectural goals have coincided.

1:25

2pAA2. Successfully persuading architects' form to follow acoustical function. David A. Conant (5655 Lindero Cyn Rd., #325, McKay Conant Hoover Inc., 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, dconant@MCHinc.com)

Architect Louis Sullivan's mantra, *Form follows function,* is no better embedded in any subdiscipline of architecture than that of building acoustics, and this is especially true in the realm of performing arts. This paper describes several examples of performing arts spaces in which MCH involved architects, often rather directly, into acoustical design. These studies were characterized, not by computer-centric tools, but rather were often most effectively and intuitively realized through simple light models that informed optimal room shaping real-time, hand-on exercises with our clients. Some solutions, especially appreciated by architects, successfully invoked indigenous or traditional forms that lent themselves beautifully to excellent speech intelligibility, musical clarity, and envelopment. This paper will discuss projects with acoustically-effective forms drawn from Native American kivas and Taoism, to Moorish tessellated patterns, to geographic forms such as southwest canyons and quarries, and more—including even "hanging chads."

1:45

2pAA3. Acoustical conflicts and synergies with energy efficient building design. Ralph T. Muehleisen (Energy Systems, Argonne National Lab., 9700 S. Cass Ave., Bldg 221, Lemont, IL 60439, rmuehleisen@anl.gov)

In the US, commercial and residential buildings consume more than 40% of the total energy and more than 75% of all electricity, contributing greatly to carbon emissions. In order to reduce both carbon emissions and reduce utility costs, buildings are being designed and constructed with much higher efficiency than even just a decade ago. Indeed, a "net-zero-ready" building is both possible and even affordable. Some design strategies can also improve building acoustics but others are detrimental. In this talk, synergies and conflicts of acoustics and energy efficient design are discussed. In particular, well-sealed building envelopes and use of high performance windows can provide both acoustic and energy benefits. However, the use of natural ventilation is at odds with acoustics. In this talk, the conflicts and synergies of these high efficiency building treatments are presented.

2:05

2pAA4. Acoustical balance between the stage and the pit in the Teatro Colón of Buenos Aires. Gustavo J. Basso (Facultad de Bellas Artes, Universidad Nacional de La Plata, Argentina, Calle 5 N° 84, La Plata, Buenos Aires 1900, Argentina, gustavobasso2004@yahoo.com.ar)

The acoustical balance between the singers and the orchestra in the well-known Teatro Colón of Buenos Aires, as an opera theater, is evaluated from several perspectives. The almost ideal balance seems to be a result of a particular combination of architectural features: among others, the shape of the horseshoe, the height and depth of the boxes at the upper levels, and the design of the proscenium and pit. In this sense, the reflections on the stage floor become significant for the higher levels. Some of the architectural causes of this acoustical behaviour have been found out from the results of opinion polls on the perceived sound by the audience, physical measurements, and the outcomes of a digital model. This paper analyses the balance between the singers, placed in various locations on the stage, and the orchestra in the pit. As it will be seen, the architectural characteristics of the theater allows the musicians to maintain the balance in almost ideal values and to preserve the spectral equilibrium and bass response through the entire hall.

2:25

2pAA5. Creative collaborations in the design of buildings. Gary W. Siebein, Hyun Paek, Marylin Roa, Keely Siebein, Jennifer R. Miller, Matthew Vetterick, and Gary Siebein (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com)

Integrating acoustical design features in rooms that fulfill acoustical, architectural, mechanical, and other functions simultaneously requires close collaboration among design team members. The efforts involved each team member understanding the other disciplines to the extent that they could creatively interpret the work. Three case studies will highlight the advantages of this type of collaboration. Full size mock-ups, computer modeling, auralizations, and other advanced design tools assist in this process. The first is a wall panel in a large museum that is used as a spatial divider, a sound attenuating return air plenum, and a surface for exhibiting artwork. A multi-function atrium was used as the central organizing feature in another museum. Design efforts explored how sounds propagated from one level to the next could be reduced. Multi-functional, layered wall assemblies that contained variable acoustic features, sound diffusing/reflecting panels, supply and return air distribution and layers to reduce sounds entering the building from the exterior were used in the renovation of a college theater. Multiple iterations of acoustical, architectural, mechanical and interior design efforts followed by intense analysis and discussion proved very rewarding for the design team with beautiful looking and sounding spaces.

2:45

2pAA6. Balancing uniform visual appearance with the need for adjustable acoustics in a concert venue: A case study. David Kahn (Acoust. Distinctions, Stamford, CT), Joshua R. Dunham (Acoust. Distinctions, 60 East 42nd St., New York, NY 10165, jdunham@ad-ny.com), Alexander Aquila, Tyler Cottrell, Jacob Ott, Nicholas Roselli, and Robert Celmer (Acoust. Prog. and Lab., Mech. Eng. Dept., Univ. of Hartford, West Hartford, CT)

An acoustician's recommendations for adjustable acoustics curtains in a concert venue sometimes comes into conflict with the architect's aesthetic vision for the space. The architectural intent is usually for patrons to have a uniform visual experience of the room, regardless of the setting of the adjustable acoustics elements. This leads to the need for architecturally interesting, acoustically transparent elements, such as perforated metal or wooden grills, that can fully or partially hide the adjustable curtains from the eyes of the audience. This paper presents a summary of the collaboration between Acoustic Distinctions and the architectural firm HGA in designing sound-transparent, patterned wood grills that enable a visually attractive and consistent architectural aesthetic, regardless of the settings of the adjustable acoustics curtains behind, in the Kracum Performance Hall at Carleton College. The wood grills—designed collaboratively by the acoustician and architect—were mocked-up and acoustical tests were carried out at the University of Hartford to document their acoustical transparency. Results from those tests will be presented. These results may provide helpful guidelines to future collaborations with architects and acousticians on the design of acoustically transparent surfaces.

3:05–3:20 Break

3:20

2pAA7. New sound system design in concert halls. Wolfgang Ahnert (ADA Acoust. & Media Consultants GmbH, Arkonastr. 45-49, Berlin D-13189, Germany, wahnert@ada-amc.eu)

In 2017, two concert halls have opened in Germany, in January Hamburg's Elb-Philharmonic Hall and in Dresden the concert hall inside the existing Kulturpalast. The presentation will report briefly about the history and development of both projects. The sound systems in both halls must serve as an announcement system but also in case of emergency as a Speech alarm system according to the German standard DIN VDE 0833-4. For both halls, the target values of Speech Intelligibility have been developed and the design approach for the needed sound systems is explained in the presentation, not only for the halls but for the partially complicated lobbies too. The excellent architectural design of the two halls led to many problems to ensure the required performance of the speaker systems, but to hide their physical visibility. An influential issue was here the interaction with the existing room-acoustic parameters in both facilities. With high reverberation values and expected high noise floors in case of emergencies the limits of needed speech intelligibility had been reached very fast. This interaction will be discussed and simulation and some measurement results are given.

Contributed Papers

3:40

2pAA8. Development and implementation of a construction noise and vibration management plan for occupied healthcare facilities. Gina Jarta and Elliott Dick (HDR, 701 Xenia Ave. South, Ste. 600, Minneapolis, MN 55416, Gina.Jarta@hdrinc.com)

Acoustics and noise management play an integral role in providing an optimum care environment for patients and staff of healthcare facilities. Indoor noise and vibration affect patient comfort and healing, clear communication, operation of sensitive imaging equipment, and overall occupant satisfaction. Construction and demolition activities often occur in occupied hospitals requiring ongoing noise and vibration sensitive operations to remain active. This case study presents the development and implementation of a construction noise and vibration management plan for a neonatal intensive care unit (NICU) expansion project. The expansion included the addition of 23 NICU beds an existing roof level, directly adjacent to an existing NICU unit, Post-Anesthesia Care Unit, operating rooms, and *In Vitro* Fertilization clinic which remained in operation during demolition and construction. The acoustical design process from establishment of baseline conditions, prediction of project-related noise and vibration, design of mitigation strategies, and development of the management plan will be presented. Monitoring data collected over the 15 month demolition and construction period will be reviewed along with project implementations strategies and lessons learned.

3:55

2pAA9. Comparative analysis of resilient and dense materials in lightweight construction for impact sound attenuation. Sean Harkin, Jacob Watrous (Eng., SoundSense, LLC, PO Box 1360, Wainscott, NY 11975, sean@soundsense.com), Bonnie Schnitta (Eng., SoundSense, LLC, East Hampton, NY), and Jennifer Scinto (Eng., SoundSense, LLC, Wainscott, NY)

Resilient underlayments are commonly utilized as a primary method of reducing footfall noise in architectural acoustics. Although resiliency is a large component of a partition's footfall performance, as well its ability to achieve higher AIIC ratings through ASTM E1007 AIIC testing, adding resiliency alone does not always address all of the frequencies which cause disturbances due to footfall. Particularly in lightweight construction, the density of the configuration is also a critical component of a successful solution. Due to the lack of sufficient mass in lightweight construction materials, successful treatment for impact noise disturbances in lightweight conditions becomes more difficult to achieve. This paper compares data with different flooring configurations in mock-up ASTM AIIC testing conditions in order to evaluate advantages and disadvantages of ANISPL (Absorption Normalized Impact Sound Pressure Level) performance in different frequencies. Resiliency and density are added in varying combinations in a wood frame construction in order to better understand their relationship to AIIC ratings and a partition's success in impact noise insulation.

4:10–4:40 Panel Discussion

Session 2pAB

Animal Bioacoustics: Plant Bioacoustics

Aaron Thode, Cochair
SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238

Simon E. Freeman, Cochair
Scripps Institution of Oceanography, 7038 Old Brentford Road, Alexandria, VA 22310

Invited Papers

1:20

2pAB1. Insect sound production and transmission in plant materials of different compositions and structures. Richard Mankin (Ctr. for Medical Agricultural and Veterinary Entomology, USDA ARS, USDA ARS CMAVE, 1700 SW 23rd Dr., Gainesville, FL 32608, Richard.Mankin@ars.usda.gov)

Insects use plants for food and shelter, and many species also have taken advantage of plant acoustical and structural characteristics to communicate for mating and social interaction over extended distances without expending significant energy. Humans have taken advantage of plant acoustical and structural characteristics to detect hidden insect infestations passively by monitoring their feeding and movement activities. This presentation reviews the characteristics of sound transmission as well as the characteristics of insect movement and feeding sounds in plant structures and products. Although insect sounds can be masked by loud background noise, different species produce sounds with particular spectral and temporal patterns that help distinguish them from background signals and from each other. Several practical applications of insect bioacoustics are discussed, including disruption of insect mating, targeting of tree pests, and monitoring of the time course of different pest management treatments.

1:40

2pAB2. Sound of wood-boring larvae and its automated detection. Alexander Sutin, Alexander Yakubovskiy, Hady Salloum, Timothy Flynn, Nikolay Sedunov (Stevens Inst. o Technol., 711 Hudson St., Hoboken, NJ 07030, asutin@stevens.edu), Hannah Nadel, and Sindhu Krishnankutty (PPQ S&T, USDA APHIS, Buzzards Bay, MA)

Stevens Institute of Technology has been investigating solutions for instrumental detection of invasive species at ports of entry. Stevens has built several acoustic systems for detection of acoustic/ vibrational signals produced by insects. This paper presents acoustic signals recorded in tests conducted in APHIS Otis Lab using tree bolts infested by Asian Longhorn Beetle, ALB and Emerald Ash Borer, EAB larvae. The analysis of the recorded sounds extracted the signal features that allowed larval classification. These features include frequencies of the generated pulses, their durations and frequencies of pulse envelops. These features showed a clear separation of ALB and EAB. For example, the main frequency of the ALB sound was in the range of 3.8–4.8 kHz, while for EAB it was between 1.2 and 1.8 kHz. A preliminary algorithm for automated insect signal detection was developed. The algorithm automatically detects pulses with parameters typical for the larva-induced sounds and rejects non-insect sound pulses. Detection is announced when the number of detected pulses for some time (5 min) exceeds the definite threshold. In the conducted test, this algorithm provided detection of a larva in all tested samples without false alarms. [This project was funded under contract with the U.S Department of Homeland Security (DHS) Science and Technology Directorate (S&T), contract HSHQDC-10-A-BOA35.]

2:00

2pAB3. Acoustic interactions between plants and animals. Michael G. Schöner and Caroline R. Schöner (Appl. Zoology and Nature Conservation, Univ. of Greifswald, Loitzer Strasse 26, Greifswald 17121, Germany, schoenerm@uni-greifswald.de)

Acoustic communication and reactions to acoustic cues are widespread and intensively studied in animals but have largely been neglected in other organisms such as plants. However, there is growing evidence for acoustic communication in plant-animal interactions. While knowledge about active acoustic signaling in plants (i.e. active sound production) is still in its infancy, research on passive acoustic signaling (i.e. reflection of animal sounds) revealed that bat-dependent plants have adapted to the bats' echolocation systems by providing acoustic reflectors, which attract mutualistic animal partners. Studies also show that plants are able to perceive sound and thus, potentially can react to animals (e.g., physiologically). Moreover, in the course of evolution plants should become acoustically more attractive to mutualistic animals that find their plant partners based on sound and less conspicuous to parasites. The current challenge is to discover further examples of plants and animals that acoustically interact with each other. Understanding the underlying proximate mechanisms and ultimate causes of acoustic communication will shed light on an underestimated dimension of information transfer between plants and animals.

2:20

2pAB4. Ultrasonic Transmission Behavior in *Posidonia oceanica* Rhizomes. Jay R. Johnson (Mech. Eng. Dept. and App. Res. Labs., The Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712, johnson.jayrichard@utexas.edu), Jean-Pierre Hermand (LISA - Environ. HydroAcoust. Lab, Université libre de Bruxelles, Brussels, Brussels Capital, Belgium), and Preston S. Wilson (Mech. Eng. Dept. and App. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

The roots and rhizomes of seagrass can form a complicated multi-phase layer within sediments. Gas channels in the rhizomes, known as aerenchyma, give rise to a complicated acoustic response. A model for acoustic propagation through such rhizomes would be beneficial for acoustic remote sensing communities. *Ex situ* measurements of the ultrasonic sound speed through the dense woody structure of the rhizomes of the seagrass *Posidonia oceanica* are presented. Ultrasonic (1–5 MHz) time-of-flight measurements were made with the rhizomes segments aligned both lengthwise and crosswise to the propagation direction of the acoustic pulse. The acoustic behaviors of plants collected from different locations in the Mediterranean are compared to each other and to the behaviors of the associated leaf blade tissues. Two measurements were also made of rhizomes before and after degassing the aerenchyma, to quantify the effects of entrained gasses on acoustic behavior. [Work supported by ONR and ONR Global.]

2:40

2pAB5. Acoustics of seagrass photosynthesis. Jean-Pierre Hermand (LISA - Environ. HydroAcoust. Lab, Université libre de Bruxelles, av. F.D. Roosevelt 50, CP165/57, Brussels, Brussels Capital 1050, Belgium, jhermand@ulb.ac.be), Jay R. Johnson (LISA - Environ. HydroAcoust. Lab, Université libre de Bruxelles, Austin, TX), Olivier Debeir (LISA - Environ. HydroAcoust. Lab, Université libre de Bruxelles, Brussels, Belgium), and Preston S. Wilson (Mech. Eng., Univ. of Texas at Austin, Austin, TX)

The paper reviews experiments in the Mediterranean that have investigated the use of low frequency sound transmission measurements (0.2–20 kHz) to study metabolism of seagrasses *in situ* in a noninvasive way and to better evaluate primary production at the scale of a meadow, for *Posidonia oceanica and Cymodocea nodosa* species. As sound interacts with the canopy and rhizosphere, the cumulative effect of multiple scattering from the aerenchymatic tissues of leaf blade, rhizome, and root changes the character of the environment impulse response. Diel variations of received energy and statistical features are due to oxygen movement which modifies the plant scattering function. During periods of high net photosynthesis, pressurization of the lacunal gas modifies the tissue mechanical properties. High internal oxygen partial pressures cause the release of oxygen into the diffusive boundary layer which can result in bubble formation on the tissue surface when local oxygen concentration exceeds solubility. *In situ* experimental results will be discussed in the light of laboratory acoustic and microscopic investigations of the tissues. [Work supported by ONR, ONR Global.]

3:00

2pAB6. Acoustic emissions by marine algae during photosynthesis. Simon E. Freeman (Naval Undersea Warfare Ctr., 6819 Duke Dr., Alexandria, VA 22307, simon.freeman@gmail.com), Lauren Freeman (Naval Undersea Warfare Ctr., Washington, District of Columbia), Giacomo Giorli (National Inst. of Water and Atmospheric Res., Honolulu, Hawaii), and Andreas F. Haas (NIOZ Royal Netherlands Inst. for Sea Res. and Utrecht Univ., Amsterdam, Netherlands)

Coastal underwater soundscapes typically contain signals from soniferous, biological processes. Identifying the sources that contribute to oftentimes acoustically complex soundscapes would facilitate a noninvasive, remote, and volumetrically integrative method to survey underwater ecological state, as has been demonstrated on land. To date the lack of knowledge on these sources has posed a challenge for extracting meaningful ecological information from underwater biological soundscape recordings. We have observed that photosynthesis by macroalgae is an acoustically active process, driven by oxygen bubbles separating from the algal surface and ringing at the Minnaert frequency. The resultant soundscapes correlate with benthic macroalgal cover across ecological gradients on shallow Hawaiian reefs during periods of daylight. Bubble size, production rate, and sound exposure level follow the concentration of dissolved oxygen, which in turn rises and falls with the availability of photosynthetically active radiation. Increased macroalgal cover results from climate related stress, overfishing and/or pollution through nutrient runoff and serves as an indicator of degradation in many underwater ecosystems. Our observation lays the foundation for signal processing methods that quickly and passively evaluate relative algal abundance in littoral waters. These findings can also be applied towards more accurate monitoring of primary production in industrial processes.

3:20–3:35 Break

Contributed Papers

3:35

2pAB7. Mating vibrational signal transmission through and between plants of an agricultural pest, the Glassy-Winged Sharpshooter. Shira D. Gordon (U.S. Dept. of Agriculture, 9611 Riverbend Ave., Parlier, CA 93648, shira.gordon@ars.usda.gov), Benjamin Tiller, James F. Windmill (Ctr. for Ultrasonic Eng., EEE, Univ. of Strathclyde, Glasgow, United Kingdom), Peter M. Narins (Dept. of Integrative Biology & Physiol., UCLA, Los Angeles, CA), and Rodrigo Krugner (U.S. Dept. of Agriculture, Parlier, CA)

The agricultural pest, glassy-winged sharpshooter (GWSS), *Homalodisca vitripennis*, relies primarily on successful vibrational communication across its home plant. Males and females engage in a vibrational duet to identify correct species, attractiveness of mate, and location on the plant. The signal produced by these animals has a dominant frequency component between 80 and 120 Hz, with harmonics spaced approximately 100 Hz apart. However, our analysis revealed that not all harmonics are present in every recorded signal. Therefore, we sought to understand how the GWSS vibrational communication signal changes over distance on the plant. We have confirmed that first, with increasing distance fewer high frequency harmonics are present. Second, at distances of only 50 cm, there is a difference in the latency of signal arrival based on the frequency, with higher frequencies arriving sooner. Finally, the animal appears to generate no airborne signal component, yet, the low frequencies are clearly detectable in neighboring plants by the signal "jumping" from leaf-to-air-to-leaf. Together, these results highlight the complexity of vibration transmission in plants and the possibility of alteration and disruption of the GWSS signal.

3:50

2pAB8. Acoustic sensing of macroalgae (kelp) for large-scale marine biomass production. Erin M. Fischell (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 77 Massachusetts Ave., 5-204, Cambridge, MA 02139, emf43@mit.edu), Andone C. Lavery, Amy Kukulya, and Timothy K. Stanton (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

An important component of the future use of large-scale offshore farms to grow macroalgae (kelp) will be remote monitoring of infrastructure, the environment, and plant health over areas so large that manual inspection is not practical. A new program, the Advanced Research Projects Agency-Energy's Macroalgae Research Inspiring Novel Energy Resources (ARPA-E MARINER), has the long-term goal of domestic energy production using biofuel derived from macroalgae. As part of that program, an integrated sensing system is being developed for unmanned underwater vehicle (UUV) monitoring of infrastructure, macroalgae growth, water properties, and associated organisms in experimental offshore macroalgae farms occupying areas square kilometers in size. A critical component of this monitoring system is acoustic sensing using a split-beam sonar system. Time-of-flight and volume backscattering data from the echosounder will be used to determine the thickness of growth and percentage volume inhabited of macroalgae. The objective is to provide a map correlated to biomass distribution variability across the farm area. Data will also be collected on aggregations of fish and zooplankton both within and outside the farm. Early results from local tests on sugar kelp will be presented, including initial research into the correlation between these acoustic data and biomass.

4:05

2pAB9. Effect of carbon content on sound speed and attenuation of sediments in seagrass meadows. Gabriel R. Venegas (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78712-1591, gvenegas@utexas.edu), Aslan Aslan, Ivy M. Hinson, Abdullah F. Rahman (School of Earth, Environ., and Marine Sci., The Univ. of Texas Rio Grand Valley, Brownsville, TX), Kevin M. Lee, Megan S. Ballard, Jason D. Sagers, Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Justin T. Dubin, and Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Globally, seagrass-bearing sediments contain 19.9 billion metric tons of carbon (C), and account for 10% of all organic C buried in the ocean each year. Protection of these C stores is vital in mitigating climate change [Fourqurean, J. W., *et al.,* Nature Geoscience 5, 505–509 (2012)]. Some sediment acoustic properties are sensitive to the presence of gas bubbles entrained in such C stores due to inherent anaerobic decomposition. Measurement of these properties could therefore provide a means to indirectly monitor C stores and overall seagrass meadow productivity. As a preliminary effort to investigate the relationship between C content and acoustic properties of seagrass-bearing sediments, cores were collected in the seagrass meadows of Lower Laguna Madre, Texas. Sound speed and attenuation from 100 kHz to 300 kHz were measured radially in 2-cm-depth increments. The cores were subsequently frozen, sliced along the same depth increments, and their C content estimated using an elemental analyzer. Acoustic properties were compared with C content along the depth of each core. The potential for sound speed and attenuation of seagrass-bearing sediments to be used as a metric for monitoring seagrass meadow productivity will be discussed. [Work supported by ONR and ARL:UT IR&D.]

TUESDAY AFTERNOON, 8 MAY 2018 GREENWAY A, 1:00 P.M. TO 4:40 P.M.

Session 2pAO

Acoustical Oceanography and Underwater Acoustics: Acoustic Seabed Characterization II

David P. Knobles, Cochair
none, KSA LLC, PO Box 27200, Austin, TX 78755

Preston S. Wilson, Cochair
Mech. Eng., Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292

Invited Papers

1:00

2pAO1. Quantifying the effect of random roughness on synthetic aperture sonar image statistics. Anthony P. Lyons (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, University of New Hampshire, Durham, NH 03824, anthony.lyons@ccom.unh.edu), Derek R. Olson (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA), and Roy E. Hansen (Norwegian Defence Res. Establishment (FFI), Kjeller, Norway)

A perturbation-theory-based model has been developed to predict the effect of random seafloor roughness on synthetic aperture sonar (SAS) image statistics. The continuous variation in scattering strength produced by a random slope field is treated as an intensity scaling on image speckle produced by the SAS imaging process. Changes in image statistics caused by roughness are quantified in terms of the scintillation index (SI). Factors influencing the SI include slope variance, geo-acoustic properties of the seafloor, the probability density function describing the speckle and the signal-to-noise ratio. Example model-data comparisons will be shown for SAS images taken off the coast of Tellaro, Italy, by the NATO Undersea Research Centre, La Spezia, Italy (now the NATO Centre for Maritime Research and

Experimentation) using the 300 kHz MUSCLE SAS system, and for data collected with the Norwegian Defence Research Establishment's 100 kHz HISAS system off the coast of northwest coast of Elba Island, Italy, in 2013. Parameter estimates obtained from high-resolution SAS data collected in these experiments will be used to illustrate the efficacy of the model and its possible use for estimating roughness parameters, such as root-mean-square slope. [Work performed under ONR Grant Nos. N00014-10-1-0051, N00014-10-1-0047, and N00014-10-1-0151, and NOAA Grant No. NA15NOS4000200.]

1:20

2pAO2. Geacoustic inversion using autonomous underwater vehicles. Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu), Venugopalan Pallayil (National Univ. of Singapore, Singapore, Singapore), and Erin Fischell (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Autonomous underwater vehicles (AUV's) are valuable for bottom geoacoustic studies because they can remain in close proximity to the seafloor, maneuver accurately along underwater tracks, and perform multi-platform operations. A REMUS 100 AUV was deployed in the recent Seabed Characterization Experiment, sponsored by the Office of Naval Research (ONR) and conducted in the New England Mud Patch Area from March to April 2017. This AUV was equipped with a sound source transmitting acoustic chirp signals from 800 to 1300 Hz and a 20 m-long digital thin line towed array (DTLA) receiving both the AUV source signals and acoustic signals transmitted from different sources. The AUV track was along a path between a moored source and a fixed receiver sled. With this experimental configuration, three different geoacoustic approaches were investigated: (1) bottom reflection inversion using the AUV source and the towed DTLA, (2) range dependent broadband acoustic inversion using the moving AUV source and the fixed receiver, and (3) range average broadband inversion using the moored source and the fixed receiver. To keep a precise time base, Chip-Scale Atomic Clocks (CSAC's) were utilized in the AUV source, the moored source and the fixed receiver, and the DTLA is synchronized to the AUV source via high-resolution acoustic interrogations. The inversion results will be presented, along with discussions for future work. [Work supported by ONR and ONRG.]

1:40

2pAO3. Estimation of sound speed and attenuation in muddy sediments using spatial coherence measurements of sound propagation. Lin Wan and Mohsen Badiey (Univ. of Delaware, 104 Robinson Hall, Newark, DE 19716, wan@udel.edu)

In the spring of 2017, a multi-national and multi-institute shallow water propagation experiment was conducted in the New England Mud Patch in order to study the acoustic properties in muddy sediments. Different types of acoustic signals (i.e., 31-g explosive charges, combustive sound source, multi-tone, and linear frequency modulation signals) deployed at various ranges, depths and azimuths were measured by one L-shaped array, one horizontal line array, and several vertical line arrays within the 30 km $\times$ 10 km experimental area. In this paper, these measured signals are first utilized to obtain the correlation coefficients for vertical coherence (VC) and longitudinal horizontal coherence (LHC), which are sensitive to the seabed geo-acoustic parameters. These spatial coherence measurements are then applied in the VC and LHC based geo-acoustic inversion algorithms [Wan *et. al*. JASA, 2016] to infer sound speed and attenuation in muddy sediments. Finally, the results from VC and LHC based geo-acoustic inversion approaches are compared with those obtained by using normal mode characteristics (e.g., modal dispersive curve with Airy phase structure, modal attenuation coefficient, and mode shapes). [Work supported by ONR.]

2:00

2pAO4. Effects of azimuthal dependent sediment layer structure on broadband acoustic propagation during Seabed Characterization Experiment in 2017. Mohsen Badiey, Lin Wan (Univ. of Delaware, University of Delaware, Newark, DE 19716, badiey@udel.edu), and John A. Goff (Inst. for Geophys., Univ. of Texas, Austin, TX)

Seabed physical properties profoundly affect acoustic energy upon interaction. The inherent structural composition of the sediment column interacting with acoustic waves has been a topic of research for decades. Besides the intrinsic physical properties that cause signal attenuation, one of the features that various sediment structures share when interacting with acoustic signals, is causing azimuthal dependence on the broadband acoustic wave propagation. This phenomenon studied in late 1990's [Badiey *et al*., JASA, 1997a, b] was revisited during the Seabed Characterization Experiment in 2017 (SBCE 2017). The sediment layer structure was obtained from the Chirp Sonar survey collected in the summer of 2015 during pilot study. These data plus CTD and bathymetric measurements are used to construct environmental input along radial tracks for acoustic field computations and range-varying wave number spectral calculations to support SBCE 2017 data analysis. The simulation results show azimuthal variability of acoustic normal modes similar to the experimental data documented in earlier studies [Badiey *et al*. JASA, 1997 and 2017]. Current results indicate that azimuthal variability of sediment physical properties is one of the causes of acoustic variability in this region. [Work supported by ONR 321OA.]

2:20

2pAO5. Ship azimuth prediction using supervised machine learning. Emma Reeves and Peter Gerstoft (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, ecreeves@ucsd.edu)

Machine learning methods are applied to noise from the R/V Endeavor across several days during the SCEx17 experiment to predict the ship's azimuth. Sample Covariance Matrices (SCMs) are formed from received pressures on two vertical line arrays (VLA1, VLA2) and one horizontal line array (SWAMI). Previously, support vector machine (SVM), feed-forward neural network (FNN), and random forests (RF) machine learning methods have accurately estimated the range of a source towed in a linear geometry in shallow water (Niu *et al*., JASA **142**, 1176–1188 (2017); Niu *et al*., JASA **142**, EL455–460 (2017)) where the training and test tracks were close together in time. In this study, we investigate the robustness of SVM and FNN for circular track prediction when the training and test tracks are taken from different days with different sound speed profiles.

2:40–2:55 Break

Contributed Papers

2:55

2pAO6. Gradient-based Bayesian geoacoustic inversion for sediment properties at the New England mud patch. Josée Belcourt, Stan E. Dosso (Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., Bob Wright Ctr. A405, Victoria, BC V8P 5C2, Canada, joseebelcourt@uvic.ca), Charles W. Holland (Appl. Res. Lab., The Penn State Univ., State College, PA), and Jan Dettmer (Dept. of Geoscinece, Univ. of Calgary, Calgary, AB, Canada)

This paper presents nonlinear Bayesian inversion of wide-angle seabed reflection-coefficient data for fine-grained/cohesive sediments recorded in the ONR Seabed Characterization Experiment at the New England mud patch. The inversion is applied to high-resolution broadband reflectivity data from a site characterized by smooth bathymetry and a thick mud layer. Since smooth, continuous gradients in seabed properties are common for low-speed mud layers, this work develops a Bayesian inversion based on a Bernstein-polynomial representation of sediment geoacoustic profiles. The gradient-based Bernstein-polynomial inversion is compared to a trans-dimensional inversion which samples over profiles defined by an unknown number of discrete seabed layers. [The research was funded by the Office of Naval Research, Ocean Acoustics Program, and the Canadian Department of National Defence.]

3:10

2pAO7. End-fire synthetic aperture sonar for seafloor volume scattering studies. Shannon-Morgan M. Steele and Anthony P. Lyons (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, ssteele@ccom.unh.edu)

Acoustic returns from seafloor sediment are comprised of scattering from both the interface and sediment volume. Although volume scattering is often the dominant mechanism, direct measurements of this component have rarely been made, if at all, due to interface roughness biasing. This bias is especially prevalent at lower frequencies where beam widths are typically 30–40 degrees. Current synthetic aperture sonar (SAS) systems are side looking and achieve narrow beam widths by coherently combining multiple acoustic pings as the sonar moves. End-fire (forward-looking) SAS would formulate a synthetic array in the same direction of travel by vertically orienting a transducer and lowering it towards the seafloor while pinging. This would create a narrow beam, significantly reducing the interface roughness bias. End-fire SAS array gains are not as substantial as conventional side-looking SAS. However, beam pattern simulations suggest the gains are still significant: a synthetic array length of 100 wavelengths can reduce a sub-bottom profiler's 40 degree beam width to 7 degrees. This talk will discuss proof of concept, motion controlled experiments performed in an acoustic testing tank and in the field.

3:25

2pAO8. Pressure and particle velocity measurements from a broad band source at ranges 1–10 km. Peter H. Dahl and David R. Dall'Osto (Appl. Phys. Lab. and Mech. Eng., Univ. of Washington, 1013 NE 40th St, Seattle, WA 98105, dahl@apl.washington.edu)

The IVAR system (Intensity Vector Autonomous Recorder) is a bottom deployed system developed for first-use in the Sediment Characterization Experiment (SCE17), conducted on the New England Mud Patch [40°28' N, 70°35' W] in spring 2017. IVAR continuously and coherently records four channels of acoustic data, three from a tri-axial accelerometer embedded in a neutrally buoyant sphere (diameter 10 cm) and one from an omnidirectional hydrophone positioned 10 cm above the centroid of the sphere positioned 1.2 m above the seafloor. The connection of these measurements to understanding seabed properties as part of SCE17 has been discussed previously. Here, emphasis is placed on documenting pressure and particle velocity signals as these quantities evolve with range from a broad band explosive source (Mk64 SUS charge). We explore the phase relation between pressure and components of particle velocity through study of active (in phase) and reactive (out of phase) intensity, as well as corresponding non-dimensional indices of the acoustic vector field, and discuss how these are influenced by propagation conditions. The work has relevance to both the ongoing geoacoustic studies from SCE17 as well as to studies on the sensitivity of fish to acoustic particle velocity generated by explosions.

3:40

2pAO9. Estimation of parameters of gassy layer in sediment in shallow water using measurement of angular dependence and spectrum of reflection coefficient of wide-band signals. Boris Katsnelson (Marine GeoSci., Univ. of Haifa, 199 Abba Khouchy Ave., Mt. Carmel, Uni of Haifa, Haifa 3498838, Israel, bkatsnels@univ.haifa.ac.il), Andrey Lunkov (General Phys. Inst., Moscow, Russian Federation), Ernest Uzhansky (Marine GeoSci., Univ. of Haifa, Haifa, Israel), and Ilia Ostrovsky (IOLR, Kinneret Lab, Migdal, Israel)

Method of estimation of gassy layer parameters in sediment (sound speed and thickness) on the base of measurement of angular and frequency dependencies reflection coefficient of wideband signals is presented. Experiments were carried out in Lake Kinneret (Israel), which is characterized by remarkable organic content in sediment producing methane bubbles. Direct chemical, biological analysis and usage of frozen cores show comparatively narrow layer (a few tens of cm) changing in dependence on season. Principal point for acoustical method is low sound speed in bubble layer (up to 300–400 m/s) and big reflection coefficient from the corresponding half-space (0.6–0.7 for vertical incident). Due to narrow gassy layer, there should be half wavelength resonances at the corresponding frequencies. In experiments, wideband signals were used (500–2500 Hz) and receiving system: single hydrophone (1 m from source) and vertical line array. Estimation of layer's parameters was carried out using measured reflection coefficient as a function of frequency and angle of reflection. Resonance frequencies were 800 Hz and 1600 Hz. The corresponding analysis of data give values of thickness 25–30 cm, with the sound speed in layer 400–450 m/c. Results are compared with experimental data obtained by direct measurements. [Work was supported by ONRG and ISF.]

3:55

2pAO10. Estimating low-frequency sediment sound speed dispersion from the horizontal coherence of the head wave excited by a light helicopter. Dieter A. Bevans and Michael J. Buckingham (Marine Physical Lab., Scripps Inst. of Oceanogr., 9500 Gilman Dr, La Jolla, CA 92093-0238, dbevans@ucsd.edu)

A series of shallow-water acoustic experiments has been conducted off the coast of southern California using a Robinson R44 helicopter as a low-frequency ($\approx$ 13–3500 Hz) sound source. The aim of the experiments was to recover the sound speed of a fine to very-fine sand sediment from the horizontal coherence of the head wave excited in the water column by the helicopter. Two hydrophones, separated horizontally by approximately 15 m and situated 0.5 m above the seabed, received the head-wave signals, allowing the coherence function to be formed over the bandwidth of the airborne source. The sediment sound speed was recovered by matching the zero crossings of the measured coherence function to those predicted from a recently developed theory of head-wave generation in shallow water. Using this technique, the dispersion in the sediment sound speed can be estimated over a frequency range extending between 27 Hz, the lowest zero crossing of the coherence function, and 3.5 kHz, the bandwidth of the source. In the middle of the frequency band, the sound speed of the sediment was estimated to be 1682 $\pm$ 16 m/s, consistent with the known sediment type. [Research supported by ONR, SMART(DOD), NAVAIR, and SIO.]

4:10

2pAO11. Sound speed profiles in the global ocean calculated from WOCE data. Mukunda Acharya (Phys. and Astronomy, Univ. of MS, 112-, 114 Chucky Mullins Dr, Oxford, MS 38655, Oxford, MS 38655, mkachary@go.olemiss.edu) and Likun Zhang (Phys. and Astronomy, Univ. of MS, University, MS)

Sound speed in the stratified ocean varies continuously with the depth. The sound speed profile reaches a minimum at a depth, that is typically 1 km in mid-latitude to form a sound channel. In this study, the CTD data

collected in the World Ocean Circulation Experiment (WOCE) are used to calculate around 6,000 sound speed profiles over the global ocean. The data of conductivity, temperature and pressure at different depths are from measurements at intervals typically in 55 km by different cruises. Sound speed is examined from the data in each cruises. By analyzing sound speed profiles, sound speed minimum and the depth for this minimum are characterized as a function of latitude ranging from 70-degree south to 70-degree north. The results show a well determined pattern. Sound speed distributions over the global ocean at various depths in shallow water are also examined.

4:25

2pAO12. Characterization of soft marine sediments using vector sensors. Oleg A. Godin (Dept. of Phys., Naval Postgrad. School, 833 Dyer Rd., Bldg 232, Monterey, CA 93943-5216, oagodin@nps.edu), Thomas J. Deal (Naval Undersea Warfare Ctr. Div. Newport, Newport, RI), and Hefeng Dong (Dept. of Electron. Systems, Norwegian Univ. of Sci. and Technol., Trondheim, Norway)

Horizontally and vertically polarized interface waves with phase speeds of 45–375 m/s were observed in the North Sea using a horizontal linear array of three-component seismometers [H. Dong *et al*., J. Acoust. Soc. Am. **134**, 176–184 (2013)]. The waves were generated by a low-frequency shear wave source on the seafloor, which operated in the 2–60 Hz band. Dispersion curves of the interface waves exhibit linear dependences between the logarithm of phase speed and logarithm of frequency, with distinct slopes at large and small phase speeds. This suggests a geoacoustic model consisting of two sediment layers with power-law dependences of the shear wave speed on depth below seafloor [O. A. Godin and D. M. F. Chapman, J. Acoust. Soc. Am. **110**, 1890–1907 (2001)]. In this paper, dispersion curves of interface waves are inverted for the shear wave speed profile in the sediments. Observations of lateral wave arrivals are used to corroborate the geoacoustic model suggested by the surface wave dispersion and constrain compressional wave speeds. Combining data for waves of different polarizations helps to identify various arrivals, check consistency of inversions, and evaluate sediment density.

TUESDAY AFTERNOON, 8 MAY 2018 — GREENWAY F/G, 1:15 P.M. TO 5:00 P.M.

Session 2pBA

Biomedical Acoustics: Using Acoustic Wave Propagation to Estimate Quantitative Material Properties of Tissue II

Matthew W. Urban, Chair
Department of Physiology and Biomedical Engineering, Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905

Contributed Papers

1:15

2pBA1. Reconstructing one-dimensional positions of spherical scatterers from overlapping echoes. Aiguo Han (BioAcoust. Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 306 N. Wright St., Urbana, IL 61801, han51@uiuc.edu)

A new method for identifying the one-dimensional positions of spherical scatterers from overlapping echoes is described. This problem is of interest because various tissues (e.g., solid tumors) can be modeled as an ensemble of discrete scatterers (e.g., tumor cells). Existing quantitative ultrasound methods are successful in estimating the scatterer size by fitting an acoustic scattering model to the backscatter coefficient estimated from the radio-frequency data. However, the scatterer positions cannot be recovered from severely overlapped echoes using such quantitative ultrasound methods alone. In the method proposed herein, the forward scattering problem is formulated based on a sparse discrete scatterer assumption. The forward problem takes into account the effects of attenuation, scattering function, transducer diffraction and system response function, all of which can be estimated or corrected using quantitative ultrasound techniques. The scatterer positions are then recovered by solving the inverse problem using compressive sensing techniques. The detection limit (distances between scatterers and number of scatterers) is investigated through numerical simulations as a function of various real-world factors such as the signal-to-noise ratio. Super resolution capability is demonstrated. Physical phantom experiments performed using single-element transducers will be presented to evaluate the method. [Support: NIH R01DK106419 and NIH R37EB002641.]

1:30

2pBA2. Estimating liver fat fraction from ultrasonic attenuation and backscatter coefficient measures in adults with nonalcoholic fatty liver disease. Lucia Albelda, Aiguo Han (BioAcoust. Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), Ethan Z. Sy (Liver Imaging Group, Dept. of Radiology, Univ. of Califiornia at San Diego, La Jolla, CA), Michael P. Andre (Dept. of Radiology and the San Diego VA Healthcare System, 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), Claude B. Sirlin (Liver Imaging Group, Dept. of Radiology, Univ. of Califiornia at San Diego, La Jolla, CA), John W. Erdman (Dept. of Food Sci. and Human Nutrition, Univ. of Illinois at Urbana-Champaign, Urbana, IL), and W. D. O'Brien, Jr. (BioAcoust. Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 306 N. Wright St., Urbana, IL 61801, wdo@uiuc.edu)

Between 80 and 100 million people in the United States suffer from nonalcoholic fatty liver disease (NAFLD). It is currently being diagnosed via MRI and biopsies; however, these tend to be expensive and painful. In this study, a Siemens 3000 ultrasound scanner was used to obtain the quantitative ultrasound (QUS) parameters [backscatter coefficient, BSC, and attenuation coefficient, AC] of 84 participants with known or suspected NAFLD (thus few uncompromised liver samples expected). Participant recruitment is continuing. Also obtained was the liver proton density fat fraction (PDFF) via MRI. AC and BSC individually were linearly correlated (~0.6) to PDFF. The data were then divided into two equally sized sets, training and test, in

order to calculate PDFF from a function of AC and BSC. Using the training set a polynomial equation was created where AC and BSC were used to estimate PDFF. It was found that a polynomial 1,2 [AC,BSC] yielded an improved correlation (0.76) when evaluated using the test set. Significant improvement is obtained to estimate PDFF using both AC and BSC as opposed to using either of these QUS parameters separately. [Support: NIH R01DK106419 and NIH R37EB002641.]

1:45

2pBA3. Direct measurement of nuclear diameter with high-frequency ultrasound (10–100 MHz) for breast cancer detection. Timothy E. Doyle (Phys., Utah Valley Univ., MS 179, 800 W. University Parkway, Orem, UT 84058-5999, Timothy.Doyle@uvu.edu), Jessica E. Carlson (Biology, Utah Valley Univ., Orem, UT), Nicole Cowan (BioTechnol., Utah Valley Univ., Orem, UT), and Garrett Wagner (Comput. Eng., Utah Valley Univ., Orem, UT)

Two high-frequency (10–100 MHz) ultrasound studies were conducted at the Huntsman Cancer Institute (Salt Lake City, Utah) on breast cancer margin and lymph node specimens from 90 patients, and produced sensitivities and specificities as high as 87.5% and 82.9%. The method used through-transmission measurements and a new parameter that correlated not only to malignant versus benign tissue, but also to various breast cancer types and other pathologies. This parameter, peak density, measures the number of peaks in the ultrasonic spectrum. The data indicated that peak density increased successively for atypical pathologies, ductal carcinomas, and lobular carcinomas as compared to normal tissue. Additional correlations were found between the distribution of peaks in the spectra and the above pathologies. This study's objective was to test the hypothesis that peak density measures the nuclear diameter of the cells in the tested tissue. A computer model of Mie scattering from a single, nucleated cell was used to calculate ultrasonic spectral peaks in the 10-100 MHz band as a function of nuclear diameter. Experiments were also conducted with agarose tissue phantoms containing polyethylene microspheres having diameters of 10-102 μm. The model and phantom results confirmed the hypothesis that peak density directly correlates to nuclear diameter.

2:00

2pBA4. Inter-sonographer reproducibility of ultrasonic attenuation backscatter coefficient measures in adults with nonalcoholic fatty liver disease. Aiguo Han (BioAcoust. Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 502 W Main St. Apt 129, Urbana, IL 61801, han51@uiuc.edu), Ethan Z. Sy (Liver Imaging Group, Dept. of Radiology, Univ. of California at San Diego, La Jolla, CA), Michael P. Andre (Dept. of Radiology and the San Diego VA Healthcare System, 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), Claude B. Sirlin (Liver Imaging Group, Dept. of Radiology, 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), and W. D. O'Brien, Jr. (BioAcoust. Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

The attenuation coefficient (AC) and backscatter coefficient (BSC) measures in the liver were found repeatable and inter-transducer reproducible in previous studies. This study assesses the inter-sonographer reproducibility of the two parameters. 56 participants (sex: 20M, 36F; age: 51.5$\pm$13.5, 24–74 yo; BMI: 32.2$\pm$5.2, 23.9–47.4 kg/m^2; 43 had MRI-PDFF: 15.6$\pm$10.4%, 0.7–41.1%) with known or suspected NAFLD each underwent two same-day ultrasound liver examinations performed by two sonographers (out of six in total) using the Siemens S3000® ultrasound system. Each examination included five RF acquisitions in separate breathholds from the right liver lobe and one from a calibrated phantom using the same settings. The AC and BSC were estimated using the reference phantom technique from each liver RF acquisition. AC and log-transformed BSC (10logBSC) are frequency-averaged within 2.6-3.0 MHz. Intraclass correlation coefficient (ICC) was used to evaluate the inter-sonographer reproducibility. Without averaging multiple intra-examination acquisitions, ICC (95% confidence interval) was 0.79 (0.65, 0.88) for AC, and 0.85 (0.74, 0.91) for 10logBSC. Averaging five intra-examination acquisitions increased ICC to 0.86 (0.77, 0.92) for AC, and 0.89 (0.81, 0.94) for 10logBSC. Theseresults show excellent inter-sonographer reproducibility of AC and BSC in patients with known/suspected NAFLD. [Support: R01DK106419 and Siemens Medical Systems.]

2:15

2pBA5. Vibro-acoustic method for detection of osteopenia and osteoporosis. Siavash Ghavami (Radiology, Mayo Clinic, College of Medicine, 200 First St. SW, Rochester, MN 55905, roudsari.seyed@mayo.edu), Mahdi Bayat, Adriana Gregory (Physiol. and Biomedical Eng., Mayo Clinic, College of Medicine, Rochester, MN), Jeremy Webb (Radiology, Mayo Clinic, College of Medicine, Rochester, MN), Max Denis (Physiol. and Biomedical Eng., Mayo Clinic, College of Medicine, Lowell, MA), Mostafa Fatemi (Physiol. and Biomedical Eng., Mayo Clinic, College of Medicine, Rochester, MN), and Azra Alizad (Radiology, Mayo Clinic, College of Medicine, Rochester, MN)

Early detection of osteopenia and osteoporosis is critical in identifying at-risk groups and manage necessary therapy. In this paper, a new noninvasive and quantitative vibro-acoustic method is proposed for detection of osteoporosis and osteopenia. In this method, we excite the bone by an ultrasound radiation force (URF) pulse. The URF pulse induces a vibration in the bone, resulting an acoustic wave that is received by a hydrophone placed on the skin. This prospective study included n = 23 volunteers. The number of volunteers with osteopenia and osteoporosis were 7, 6, respectively, and the number of volunteers with normal bone was 10. URF excitations were applied at four equi-distance points on the bone. The resulting acoustic signals were used for wave velocity estimation based on cross-correlation technique. In general, the hydrophone signal may be modeled as a superposition of different modes of bone vibration. Based on this model, we used variational mode decomposition (VMD) as an efficient technique to decompose the received signal into an ensemble of band-limited intrinsic mode functions. Statistical analysis demonstrates that osteopenia and osteoporosis bones can be differentiated from normal bone with $p< 0.001$ using Wilcoxon rank-sum test on estimated velocity of the modes with maximum power.

2:30

2pBA6. Comparison of conventional transducer and refracto-vibrometry measurements of ultrasonic transmission parameters from heel bones. Thomas M. Huber, Matthew R. Mehrkens, Benjamin A. Rorem (Phys., Gustavus Adolphus College, 800 W College Ave., Saint Peter, MN 56082, huber@gac.edu), Matthew T. Huber, and Brent Hoffmeister (Phys., Rhodes College, Memphis, TN)

Ultrasonic measurements of the heel bone (calcaneus) are used commonly for osteoporosis screening. Ultrasound pulses that pass through the calcaneus are detected with a receiving transducer. Parameters, such as normalized broadband ultrasound attenuation (nBUA) and speed of sound (SOS) calculated from the wave forms detected by the receiving transducer, are utilized to assess bone health. In the current study, refracto-vibrometry (RV), an interferometric method for optically measuring ultrasound, was compared with conventional transducer measurements of ultrasound transmission *in-vitro* through a human calcaneus sample. The measurement beam from a Polytec PSV-400 scanning laser Doppler vibrometer was directed through a water tank towards a stationary retroreflective surface. Acoustic wave fronts (density variations) which pass through the ~50μm diameter measurement laser cause variations in the integrated optical path length. The signals detected by the vibrometer at numerous scan points were used to determine the time evolution of ultrasonic wave fronts. Because there is not a physical receiving transducer, the RV measurements do not experience the same frequency response and reflection artifacts of conventional transducers. Parameters such as nBUA and SOS were calculated at multiple RV scan points, and the results were compared to parameters measured using a conventional ultrasound transducer.

2p TUE. PM

2:45

2pBA7. A continuous power law wave equation for biomedical ultrasound. James F. Kelly, Mark M. Meerschaert (Dept. of Statistics and Probability, Michigan State Univ., 1. East Lansing, MI 48824, kellyja8@stt.msu.edu), and Robert McGough (Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI)

The attenuation coefficient in most biological media is a power law with respect to frequency. Measurements indicate that the power law exponent y is near or exactly one for many classes of tissue. In J. Acous. Soc. Am. [**124** 2861–2872 (2008)], a power law wave equation (PLWE) was proposed to model this power law dependence using Riemann-Liouville fractional derivatives. The PLWE, like most other fractional calculus models for attenuation in the time-domain, is invalid for $y = 1$ due to a discontinuity in the phase velocity. To address this problem, a continuous power law wave equation (CPLWE) is proposed that is valid for all power law exponents between zero and two. The CPLWE utilizes the recently developed Zolotarev fractional derivative of order y, which is a nonlocal operator even for $y = 1$. The 3D Green's function for the CPLWE that is valid for homogeneous media is derived using stable probability density functions in the Zolotarev M-parameterization. Solutions to the CPLWE that are valid for inhomogeneous media are then expressed as solutions to the wave equation subordinated to an inverse stable density. These subordinated Green's function solutions transition smoothly between $y<1$ and $y>1$ power laws, which facilitates estimation of attenuation parameters in biological media with y close to one.

3:00–3:15 Break

3:15

2pBA8. Power law attenuation of shear waves in fractal media. Sverre Holm (Dept. of Informatics, Univ. of Oslo, Gaustadalleen 23B, Oslo N 0316 Oslo, Norway, sverre@ifi.uio.no) and Ralph Sinkus (Div. of Imaging Sci. and Biomedical Eng., Kings College London, London, United Kingdom)

There are two mechanisms for attenuation of mechanical waves: absorption and multiple scattering. We explore the second mechanism for shear waves in the context of MR elastography. The theory for attenuation was first given in the seismic field (O'Doherty & Anstey, Geophys. Prosp. 1971). Later phase was included, and the mean field concept was applied (Banik *et al.*, Geophys. 1985). Then, a more rigorous theory which also deals with random fractals was developed (Sølna, SIAM J ApplMath, 2003; Garnier & Sølna, Multiscale Model Simul, 2009). We have developed this theory further by allowing fractality over a limited length scale and also shown that it gives results that are indistinguishable from those of power law absorption models like the fractional Kelvin-Voigt model (Holm & Sinkus, JASA, 2010). This makes it a challenge to understand whether attenuation in a real medium is due to one or the other mechanism or both. Measurement of dispersion has been done for polystyrene microspheres embedded in agarose which is nearly lossless up to 1000 Hz. The phase velocity has a power law variation with exponent up to $d = 0.24$ (Lambert *et al.*, PRL 2015). According to the theory this is consistent with a fractional dimension which is d less than the Euclidean dimension.

3:30

2pBA9. Subharmonic generation by shear waves in a 1D resonator formed with a relaxing material. John M. Cormack and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, jcormack@utexas.edu)

The low shear moduli of soft elastic media such as rubber and tissue facilitate the excitation of shear waves that exhibit significant finite-amplitude effects. Measurements performed by Andreev *et al.* [Acoust. Phys. **57**, 779 (2011)] of the amplitude-dependent response of a 1D shear-wave resonator formed with a rubber-like material revealed reasonable agreement with numerical simulations based on a monorelaxing material model. At the previous ASA meeting the present authors developed an augmented Duffing equation as a model for plane shear waves in a resonator formed with a monorelaxing material that is shaken at one end and free at the other [J. Acoust. Soc. Am. **142**, 2723 (2017)]. The focus of the previous work was on the response at drive frequencies near the lowest resonance. Here the augmented Duffing model is used to investigate subharmonic generation associated with a drive frequency of approximately three times that of the lowest mode. Conditions on the amplitude and frequency of excitation for which subharmonic generation can occur are obtained from the augmented Duffing equation. Results obtained from the augmented Duffing equation are compared with direct numerical solutions of a nonlinear wave equation. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

3:45

2pBA10. Parametric phantom study of shear wave velocity image reconstruction. Matthew W. Urban (Dept. of Radiology, Mayo Clinic, 200 First St. SW, Rochester, MN 55905, urban.matthew@mayo.edu) and Jorge Racedo (Dept. of Biomedical Eng. and Dept. of Phys., Universidad de los Andes, Bogota, Colombia)

Shear wave elastography is being increasingly used for multiple applications including investigation of diffuse disease in the liver or kidney and malignancies or neoplasms in the breast and thyroid. It is important to provide robust and accurate reconstructions of shear wave velocity (SWV) for imaging purposes to provide reliable data for diagnosis. One such approach was a two-dimensional method for estimating SWV (Song, *et al.*, Ultrasound Med. Biol. 40, pp. 1343–1355, 2014). This method uses a weighted average using a large window and smaller patches to perform time-domain cross-correlations for the calculation of the SWV. We also implemented a spatial sigmoid weighting to combine reconstruction results from independent reconstructions after directional filtering. We used a Verasonics system equipped with a linear array transducer to conduct a parametric study in phantoms to investigate the effects of the window and patch on image reconstruction metrics. Measurements were made in homogeneous phantoms and in a phantom with cylindrical inclusions of different diameters. Larger windows and patches typically yielded low variability but did not always produce the highest contrast. This investigation is the basis for an adaptive method for reconstructing SWV in soft tissues and choosing optimal parameters for the SWV reconstruction.

4:00

2pBA11. Quantifying nonlinear elasticity modulus of tissue-like solids using acoustic radiation force. Danial Panahandeh-Shahraki, Bojan B. Guzina (Dept. of Civil, Environ. and Geo- Eng., Univ. of Minnesota, 500 Pillsbury Dr. SE, Minneapolis, MN 55414, panah004@umn.edu), Viksit Kumar (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine & Sci., Rochester, MN), Matthew W. Urban (Dept. of Radiology, Mayo Clinic College of Medicine & Sci., Rochester, MN), Randall R. Kinnick (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine & Sci., Rochester, MN), Siavash Ghavami, Azra Alizad (Dept. of Radiology, Mayo Clinic College of Medicine & Sci., Rochester, MN), and Mostafa Fatemi (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine & Sci., Rochester, MN)

As shown in a recent study, estimation of the magnitude of the acoustic radiation force (ARF) in isotropic tissue-like solids is the key to measuring a third-order nonlinear modulus of elasticity (C)—responsible for coupling the shear and volumetric constitutive responses of a material. Due to experimental limitations, a direct measurement of the ARF in tissue is however difficult. To overcome this problem, we deploy 3D elastodynamic finite element (EFE) simulations of the ARF experiment in tissue-mimicking phantoms to estimate (i) the "background" shear modulus from the phase of the induced shear waves; (ii) the ARF magnitude from the shear wave amplitude, and (iii) the modulus C from the knowledge of the ARF. In this way, by interpreting the ARF-generated shear waves through the prism of 3D-EFE simulations, we facilitate local estimation of the nonlinear C modulus with spatial resolution equaling the size of the focal region. We call this method the C-Elastography (CE). In this study, the CE technique is applied to gelatin phantoms containing an agar-based inclusion. The profiles of the C-modulus in phantoms tested demonstrate good correlation with the phantom geometry, showing marked (and sharp) C-contrast at push points acting on the inclusion. [Acknowledgment: NIH-Grant EB23113.]

4:15

2pBA12. Finite element analysis of lung ultrasound surface wave elastography. Boran Zhou and Xiaoming Zhang (Radiology, Mayo Clinic, 321 3rd Ave. sw, Rochester, MN 55902, Zhou.Boran@mayo.edu)

Interstitial lung disease manifested as fibrotic and stiffened lung tissue, resulting in symptoms such as dyspnea and may lead to respiratory failure. We recently developed a lung ultrasound surface wave elastography for evaluating the elastic properties of superficial lung tissue. In this abstract, we present a finite element analysis to investigate the effects of pleural effusion on the shear wave propagation of superficial lung tissue. The muscle, pleural effusion, and lung tissue were simulated by a 2D planar model of an infinite elastic medium with densities for different materials. A nearly incompressible, linear elastic model was used to model muscle, pleural effusion and lung. The model was excited using a line source on the muscle surface. Harmonic excitations were performed at 100, 150 and 200 Hz with duration of 0.1 s. The boundaries of lung and muscle were attached to an infinite. The mesh of muscle, pleural effusion, and lung were constructed using linear quadrilateral elements. The infinite region was meshed by infinite elements (type CINPE4). The dynamic responses of the tissue model to the excitations were solved by the ABAQUS explicit dynamic solver with automatic step size control. Insignificant difference in shear wave speeds of lung tissue at different excitation frequencies was found in both simulations with and without taking into account the pleural effusion, indicating that pleural effusion does not affect the measurements of shear wave speeds in ultrasound elastography.

4:30

2pBA13. Pleural fluid may pose no measurable effect on lung stiffness in lung ultrasound surface wave elastography. Jinling Zhou and Xiaoming Zhang (Dept. of Radiology, Mayo Clinic, 1-24 Medical Sci. Bldg., 321 3rd Ave. SW, Rochester, MN 55902, zhou.jinling@mayo.edu)

Lung ultrasound surface wave elastography (LUSWE) was developed as a non-invasive tool for assessing pulmonary fibrosis. In this research, the effect of the fleural fluid on surface wave speed measurement of the lung is studied. Low-cost sponges [J. Ultrasound Med. 2017; 36, 2133], as good lung phantoms, were used in this study. The ultrasound transmission gel was used to simulate the pleural fluid. A harmonic vibration was generated at sponge surfaces using a shaker at 100, 150, and 200 Hz. The sponge surface wave speeds were measured with a 6 MHz ultrasound probe for both with and without a thin layer gel. The ultrasound features of A and B-lines were clearly observed in every sponge model in our experiment. The surface wave speeds for the sponges with the gel were indistinguishable from those without the gel for each frequency. The surface wave speeds were faster at higher excitation frequencies which is consistent with our patient study results. It was conclude that the pleural fluid may not affect the lung surface wave speed measurement based upon our lung phantom sponge models.

4:45

2pBA14. A deep neural network method for analyzing lung density using lung ultrasound surface wave elastography. Boran Zhou and Xiaoming Zhang (Radiology, Mayo Clinic, 321 3rd Ave. sw, Rochester, MN 55902, Zhou.Boran@mayo.edu)

Lung density is directly associated with lung pathology. Computed Tomography (CT) evaluates lung pathology using the Hounsfield unit (HU) but not directly the lung density. We recently developed a lung ultrasound surface wave elastography (LUSWE) technique for measuring the surface wave speed of superficial lung tissue. The objective of this study is to analyze lung density of superficial lung tissue using deep neural network (DNN) and wave speed measurements obtained from LUSWE. The synthetic training dataset (7,88,000 in total) in terms of wave speeds of superficial lung tissue at different excitation frequencies from LUSWE, viscoelasticity and density of lung tissue was used to train the DNN. DNN is composed of 3 hidden layers of 1024 neurons and trained for 10 epochs with a batch size of 4096 and a learning rate of 0.001 with Adam optimizer. Test dataset in terms of wave speeds at different excitation frequencies (100, 150, and 200 Hz) as well as elasticity was used to predict the lung density and evaluate its accuracy. The obtained results showed that predictions matched well with test dataset (validation accuracy is 0.997). This method may be useful to analyze lung density based on the LUSWE measurements.

TUESDAY AFTERNOON, 8 MAY 2018 GREENWAY D, 1:00 P.M. TO 2:15 P.M.

Session 2pEA

Engineering Acoustics and Physical Acoustics: Miniature Acoustic Transducers II

Vahid Naderyan, Chair
Physics/National Center for Physical Acoustics, University of Mississippi, NCPA, 1 Coliseum Drive, University, MS 38677

Contributed Papers

1:00

2pEA1. Artificial design of open-back headphone based on ventilation structure optimization method. Yu-Ting Tsai (Feng Chia Univ., No. 100 Wenhwa Rd., Seatwen, Taichung City 40724, Taiwan, yuttsai@fcu.edu.tw) and Jin H. Huang (Feng Chia Univ., Taichung City, Taiwan)

The computer-aided analysis of open-back headphone based on finite/bound element method, lumped parameter model, and transduction equations have been investigated for decades. In a serious attempt to reach the optimal design of headphone, various aspects of conceptualization, design, modeling, and simulation are investigated for improvement of headphone in this work. The standardized construction process of the three-dimensional image of the human ear to establish the open-back architecture for the headphone is given. For improving the sound pressure level response of the headphone, the parameter design method is proposed through the optimized dimension modification of special open-cavity structure. All results of studies of the simulated responses are compared with experimental data and have verified the veracity of the simulations. The present work has concluded the feasibility of adjusting the model and parameters for customizing the resulting response for headphone's varied applications.

1:15

2pEA2. Frequency response simulation of the hybrid dual driver earphone using equivalent circuit model. Yu-Ting Tsai (Feng Chia Univ., No. 100 Wenhwa Rd., Seatwen, Taichung City 40724, Taiwan, yuttsai@fcu.edu.tw)

By combining the multiple drivers into an insert earphone, the special filtering of hybrid dual drivers by earphone structure design can be applied to increase efficiency and the overall output level. In general, the back of the earphone structure lays a balanced armature driver for the mid to high frequency reproduction and a dynamic driver in front cavity creates deeper bass. For this distinctive structure of the earphone, this study proposed a frequency response simulation model using the equivalent circuit model. For the sound transmission between the two ports' structure, the correction factors for obtaining the transmitted impedance of special filter are proposed. The simulated responses shows that a good comparison between the measured result and the equivalent circuit simulation. As a result, the study has significant capacity to estimate the desired frequency response of a hybrid dual driver earphone.

1:30

2pEA3. Miniature implantable low noise piezoelectric diaphragm sound sensor. Chuming Zhao, Alison Hake (Mech. Eng., Univ. of Michigan, 2350 Hayward, Ann Arbor, MI 48109, chumingz@umich.edu), Wang-Kyung Sung (Vesper Technologies Inc., Boston, MA), and Karl Grosh (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Piezoelectric diaphragm sound sensors must be miniaturized and environmentally toughened for use in biomedical applications. We study two disparate applications of the sensor, first as the front end of a completely implantable cochlear implant and the second as an intracardiac pressure sensor. The sensor, which is fabricated using micro-electrical-mechanical-system (MEMS) techniques, consists of a circular diaphragm with at least one aluminum nitride (AlN) based piezoelectric layer backed by an air-filled cavity. The sensors are designed to have a lower input referred noise (IRN) compared to other miniaturized piezoelectric diaphragm hydrophones with similar diaphragm size via structural design and electrode area optimization. Sensors are fabricated with varying diaphragm diameters from 300 to 450 μm to test the effect of residual stresses due to the manufacturing process on the frequency bandwidth and IRN. The sensor works in air and underwater with the same sensitivity, but the frequency bandwidth is reduced underwater compared to that is in air because of the water mass loading. A finite element analysis model using COMSOL is built to study the in air and underwater sensing process and the IRN.

1:45

2pEA4. Experimental assessment of miniature accelerometers designed for sensing middle ear motion. Alison Hake, Chuming Zhao (Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109, aehake@umich.edu), Wang-Kyung Sung (Vesper Technologies, Medford, MA), and Karl Grosh (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Sensing human middle ear motion requires miniature, low-noise sensors to detect low-amplitude vibration of the ossicle bones to replace the external microphone in a cochlear implant system. Specifically, a 35 $\mu g/\sqrt{Hz}$ sensor resolution is needed measure motion at 500 Hz (Young *et al.*, 2012). Motivated by sensing middle ear motion without irreversible alteration to the ossicular chain, a small, piezoelectric accelerometer is considered. Our detailed low-frequency analytical model is used to identify promising micromachined, piezoelectric bimorph cantilever accelerometer designs. We fabricated a first generation of accelerometers using a conservative microfabrication approach, taken to enhance the yield of devices rather than to minimize the input referred noise (IRN). The mathematical model was validated by voltage actuation testing of these accelerometers. The device sensitivity to acceleration is tested, and the resulting IRN is desired to assess the viability of the sensor for the middle ear motion application. The first generation experimental results are compared to the mathematical model to determine the accuracy of the model and the efficacy of the fabrication methodology. This knowledge will influence future sensor designs optimized for minimal IRN.

2:00

2pEA5. A new approach for an FE-model optimization of a MEMS piezoelectric accelerometer for implantable hearing devices. Zargos Neves Masson, Andre L. Gesing, Matheus de Lorenzo (Mech. Eng., Federal Univ. of Santa Catarina, Av. Madre Benvenuta,40, ap 302, Florianopolis, Santa Catarina 88036500, Brazil, zargosm@gmail.com), Stephan Paul (Mech. Eng., Federal Univ. of Santa Catarina, Joinville, Brazil), and Julio A. Cordioli (Mech. Eng., Federal Univ. of Santa Catarina, Florianópolis, SC, Brazil)

Totally implantable hearing devices have been proposed as a solution to mitigate the constraints related to the presence of external elements in traditional hearing devices. This work reports on a novel approach for the development of a MEMS piezoelectric accelerometer as an implantable sensor for hearing devices. Traditionally these sensors are designed for maximum sensitivity; however, the primary bottleneck for this type of transducer is the internal noise level. Two differential evolution optimization routines were developed. The first used sensitivity maximization whereas the second seeks minimization of the equivalent input noise (EIN). Both methods were applied to different designs of Lead Zirconate Titanate (PZT) MEMS piezoelectric accelerometer in the frequency range of 250 Hz to 8 kHz with layer thickness varying from 0.1 μm to 1 μm. In the latter approach, the sensor's acceleration noise was estimated analytically, considering capacitance and charge response acquired through finite element modeling (FEM) previously validated. Acceleration noise was converted to EIN, in sound pressure level (SPL), through an FE-model of the middle ear considering the sensor coupled at the umbo. Preliminary results indicate that using EIN as optimization goal, opposed to sensitivity, leads to a higher performance over a broader bandwidth.

Session 2pNS

Noise and ASA Committee on Standards: Effects of Natural Soundscapes on Recreation Areas

David Braslau, Cochair
David Braslau Associates, Inc., 6603 Queen Ave. S, Suite N, Richfield, MN, MN 55423

Kurt M. Fristrup, Cochair
Natural Sounds and Night Skies Division, National Park Service, 1201 Oakridge Drive, Suite 100, Fort Collins, CO 80525

Chair's Introduction—1:00

Invited Papers

1:05

2pNS1. Conflicting sounds in natural recreation areas—An historical Minnesota perspective. David Braslau (David Braslau Assoc., Inc., 6603 Queen Ave. S, Ste. N, Richfield, MN 55423, david@braslau.com)

Minnesota has long seen battles between conflicting activities in natural recreation areas. Snowmobiles were first introduced in the 1960 by two Minnesota firms. By the 1970s physical and noise impacts had increased and a state committee to address trail uses and conflicts was established. In the same time period, a large scale environmental impact study on effects of copper-nickel mining on the BWCAW and national forest was initiated. A noise model was developed by Moorhead State staff that compared mining noise with natural noise levels. This issue rested until 2007 when exploration drilling started 24/7 in the Superior Nation Forest and near the BWCAW. Extensive monitoring and modeling efforts were undertaken by the companies and the Forest Service and limits established on acceptable noise levels. The snowmobile noise problem has not gone away with periodic controversy and assessment. To the south, the Saint Croix National Scenic Riverway (NPS) and the Minnesota Valley National Wildlife Refuge (USFWS) have both addressed potential noise intrusion from adjacent sand and gravel mining activities. Of course, the BWCAW was the center of long term disputes over noise, with motors now limited to only a few lakes and low altitude aircraft flights prohibited.

1:25

2pNS2. Management considerations for noise impacts on the superior National Forest and Boundary Waters Canoe Area Wilderness. Peter Taylor and Ann Schwaller (Superior National Forest, Forest Headquarters, 8901 Grand Ave. Pl., Duluth, MN 55808, prtaylor@fs.fed.us)

The USDA Forest Service-Superior National Forest manages the Boundary Waters Canoe Area Wilderness (BWCAW) and multiple use lands located outside the Wilderness. For the BWCAW, the Forest Service is responsible for preserving wilderness character per Section 4b of the 1964 Wilderness Act. Accordingly, Forest Service managers need to understand the effects of human-generated sound on the soundscapes of the Wilderness when making decisions on if, where, and how to permit management activities to occur. Human-generated sound that may affect the Wilderness has been a consideration in management decisions for timber harvest, snowmobiling, ATVing, minerals exploration, and mining. Further, human-generated sound may also be an issue for recreation users or residents located on multiple use lands outside the Wilderness. We discuss the ongoing need for scientific information to inform decision making on this issue. We also discuss our monitoring efforts and data and analysis needs.

1:45

2pNS3. Modeling and mapping aircraft noise in National Parks. Amanda Rapoza (Volpe National Transportation Systems Ctr., U.S. Dept. of Transportation, 55 Broadway, RVT-41, Cambridge, MA 02142, amanda.rapoza@dot.gov), Brent Lignell, Kurt M. Fristrup (Natural Sounds and Night Skies Div., National Park Service, Fort Collins, CO), and Aaron L. Hastings (Volpe National Transportation Systems Ctr., U.S. Dept. of Transportation, Cambridge, MA)

Commercial air tours are a common source of noise within many national parks, affecting natural and cultural resources as well as visitor experience. To facilitate interactive modeling and mapping of air tour noise exposure, the Volpe National Transportation Systems Center and National Park Service (NPS), in cooperation with the Federal Aviation Administration (FAA), have created an Aviation Noise Analysis and Mapping Tool. This tool contains a library of park noise maps generated by modeling single aircraft operations on routes flown by that aircraft. The creation of this library is computationally intensive and requires considerable expertise to properly configure and run the models. With this tool and library, a broad range of users can interactively explore composite noise exposure generated by air tour traffic scenarios. Composite noise exposure is calculated by summing, across all aircraft and route combinations, the product of the number of flights on each route and the noise generated by each aircraft on that route. The use of FAA noise models

ensures consistency with other U.S. environmental analyses, use of state-of-the-art algorithms, and access to updated aircraft noise databases. The traffic aggregation component can be modified to incorporate specialized interpretations of noise exposure for NPS applications.

2:05

2pNS4. Forecasting increases in recreational value that would result from restoration of natural soundscapes in National Parks. Kurt M. Fristrup (Natural Sounds and Night Skies Div., National Park Service, 1201 Oakridge Dr., Ste. 100, Fort Collins, CA 80525, kurt_fristrup@nps.gov), Megan F. McKenna, and Daniel J. Mennitt (Natural Sounds and Night Skies Div., National Park Service, Fort Collins, CO)

U. S. National Parks are justly celebrated for the superlative quality of their scenic and cultural resources. The Grand Canyon Enlargement Act of 1975 identified natural quiet as an important resource and value. These considerations are two motivations for extensive monitoring of acoustic environments in national park units, and development of models that predict sound levels throughout the system. These models also estimate the spatial distribution of noise of noise exposure. Related social science research has documented the effects of noise on the quality of visitor experience in parks, including the accessibility of wildlife for public viewing. Surveys of park visitors and the American public also document broad support of park efforts to reduce noise. Collectively, this research provides a quantitative framework for estimating the increases in the quality of visitor experience that could be realized through reductions in noise.

2:25–2:40 Break

Contributed Papers

2:40

2pNS5. "Listening" Crowdsourced Sound. Yalcin Yildirim (Urban Planning and Public Policy, Univ. of Texas at Arlington, 601 W Nedderman Dr #203, Apt. 933, Arlington, TX 76019, yalcin.yildirim@mavs.uta.edu)

Bearing in mind the influence of internet based instruments for crowdsourcing, landscape architecture, urban planning, and urban design fields are progressively applying these tools to obtain better notions and alternatives from the community. Such instruments generally provide considerable data about what community desires. In spite of this fact, to the best of our knowledge, crowdsourced intelligence in landscape architecture and urban planning has been studied limited. This research concentrates on University of Texas at Arlington campus soundscape in the heart of 6 million populated Dallas-Fort Worth metroplex to examine the opportunities and applications of performing crowdsourced data. To do this, research team applied mixed methods, the study evaluates the opinions of campus users and determines in which aspects those ideas can be connected to soundscape patterns. After investigating the interviews of campus users, the study integrated the information of perceptions about opinions at the end of the research. The findings emphasizes the challenges, limitations, and opportunities in regard to the landscape architecture, urban planning, and urban design disciplines. It is noteworthy that several circumstances have implications on applicability of crowdsourced knowledge on soundscape.

2:55

2pNS6. A basic research on expression of loudness and orientation of sound source in soundscape by onomatopoeia on pictures. Takeshi Akita (Dept. of Architecture, School of Sci. and Technol. for Future Life, Tokyo Denki Univ., 5 Senju-Asahi-cho Adachi-ku, Tokyo 1208551, Japan, akita@cck.dendai.ac.jp), Takaaki Koga (Utsunomiya Univ., Utsunomiya, Japan), Naoko Sano, and Ayako Matsuo (Tokyo Denki Univ., Adachi-ku, Tokyo, Japan)

When soundscape is described with words under common acoustic environment, it is usual that the name of the sound source that indicates their category or species is noted. Generally, such kind of description cannot express loudness and orientation of sound source. However, the information of loudness and orientation of the sound source is important, so it is necessary to note them easily at a time. In the present research, it is investigated whether description of soundscape using onomatopoeia on pictures to express loudness and orientation of sound sources is effective or not. Two psychological experiments are carried out to reveal the availability of this description method. Results show that onomatopoeia can present the type of source, and that it can simultaneously express variety of loudness of sound source and noise. It is suggested that the size and number of onomatopoeia described on pictures of the place where noise exists represents loudness of sound and noise, and that the variety of size of onomatopoeia can show the difference of sound pressure level of 20 dB.

3:10

2pNS7. Social investigation on passersby perception of urban fountain sounds. Laura Velardi (LISA - Environ. HydroAcoust. Lab, ULB- Université libre de Bruxelles, av. F.D. Roosevelt 50, Brussels 1050, Belgium, lvelardi@ulb.ac.be) and Jean-Pierre Hermand (LISA - Environ. HydroAcoust. Lab, ULB- Université libre de Bruxelles, Brussels, Brussels Capital, Belgium)

Running water sounds are among the most pleasant sounds to human ears. Beside natural environments, people experience these sounds in urban environments thanks to fountains. The influence of fountain sounds on passersby has little been explored, especially *in situ*. This paper presents results from social surveys conducted in Rome and Brussels' squares for monumental and contemplative fountains as well as modern and participative fountains. Over 70 % of survey respondents qualified the fountain sounds as the most interesting feature of the place, even though the questionnaire did not make any explicit reference to acoustics. For each case study, a comprehensive set of timbre descriptors was applied to audio recordings in the fountain surroundings to characterize sound pleasantness in relation to people's impressions. Beyond their marked interest for the fountain soundmark, the survey identified people's preferences in terms of arrangement of fountain surroundings. Such information can be useful to forthcoming urban acoustic design, currently lacking any in depth observation about pedestrian land use related to urban sounds.

3:25

2pNS8. Noise dependent coherence-super Gaussian based dual microphone speech enhancement for hearing aid application using smartphone. Nikhil Shankar, Gautam Shreedhar Bhat, Chandan Karadagur Ananda Reddy, and Issa M. Panahi (Elec. and Comput. Eng., Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080, nxs162330@utdallas.edu)

In this work, the coherence between speech and noise signals is used to obtain a Speech Enhancement (SE) gain function, in combination with a Super-Gaussian Joint Maximum a Posteriori (SGJMAP) single microphone SE gain function. The proposed SE method is implemented on a smartphone that works as an assistive device to hearing aids in real-time. Although coherence SE gain function suppresses the background noise well, it distorts the speech. In contrary, SE using SGJMAP improves speech quality with the introduction of musical noise, which we contain by using a post filter.

The weighted union of these two gain functions strikes a balance between noise suppression and speech distortion. A "weighting" parameter is introduced in the derived gain function that can be controlled by the smartphone user based on different background noises and their comfort level of hearing, which proves the practical usability of the developed SE method implemented on the smartphone. Objective and subjective measures of the proposed method show significant improvements in both quality and intelligibility compared to the state of the art SE techniques considered in this work for several noisy conditions at the signal to noise ratio levels of -5 dB, 0 dB, and 5 dB.

TUESDAY AFTERNOON, 8 MAY 2018 GREENWAY J, 1:00 P.M. TO 4:10 P.M.

Session 2pPA

Physical Acoustics: Infrasound for Global Security II

Philip Blom, Chair
Los Alamos National Laboratory, Los Alamos National Laboratory, PO Box 1663, Los Alamos, NM 87545

Contributed Papers

1:00

2pPA1. Acoustic waveform inversion and uncertainty analysis for explosion yield estimation. Keehoon Kim and Arthur Rodgers (Geophysical Monitoring Program, Lawrence Livermore National Lab., 7000 East Ave., L-103, Livermore, CA 94550, kim84@llnl.gov)

Waveform inversion techniques are widely used to extract source information from acoustic signals. The methods exploit full-waveform information and provide further constraints on source inversion than other parameter-based inversions. Infrasound waveform inversion was recently applied to chemical explosions and showed promising results for the determination of explosion yield. The method can improve the accuracy of the estimated yield by using full 3-D numerical Green's functions and incorporating the propagation path effects into the inversion. However, the inversion results are often given without any uncertainty estimation, which is critical for a post-detonation forensic analysis. In this study, we evaluate the yield estimation errors associated with the uncertainty in the atmospheric representation. An ensemble of atmospheric realizations is obtained by adding random turbulence to an atmospheric model and the uncertainty of inversion solution depending on atmospheric variation is quantified. In addition, different atmospheric models derived from local weather data or numerical weather prediction models are validated for yield estimation by using a series of chemical explosion data.

1:15

2pPA2. Estimation of explosive yield using regional distance infrasound. Philip Blom, Fransiska K. Dannemann, and Omar Marcillo (Los Alamos National Lab., Los Alamos National Lab., PO Box 1663, Los Alamos, NM 87545, pblom@lanl.gov)

Infrasound signals from large above-ground explosions are known to propagate to significant distances while remaining at observable levels due to a combination of the large source energy for such events and decreased absorption for acoustic energy at lower frequencies. Previously, propagation-based, stochastic path geometry and travel time models have been used to improve estimates of location and time for infrasonic sources with promising results. A similar approach has been applied to develop stochastic transmission loss models for infrasonic signals. The resulting models have been utilized in a Bayesian framework to compute a near-source estimate of the acoustic signal, which can then be used to compute a probability distribution for possible explosive yields of the source. Details of the propagation models and yield estimation framework will be presented along with results for application to a number of explosive events.

1:30

2pPA3. Improvements to infrasonic detection capabilities through application of a generalized least squares beamformer. 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)

A common challenge in seismoacoustic research and analysis is detection of low signal-to-noise ratio (SNR) signals in a variable and often coherent background. Previously, the application of an adaptive F-detector to infrasonic data has successfully reduced false detection rates attributed to coherent noise across array elements; however, the detector is applied post-processing after analysis using a standard (Bartlett) beamformer, which raises the detection threshold and can lead to missed detections. Application of a generalized least squares (GLS) beamformer can enhance processing of transient infrasonic signals, particularly in the presence of correlated noise, by adaptively accounting for the background noise characteristics. The GLS beamformer enhancement leads to improved capabilities for accurately detecting low amplitude infrasonic events of interest that would not normally produce a sufficiently high F-statistic to be declared a detection. A statistical analysis of GLS beamformer performance in a variety of noise environment will be presented, along with the statistical significance of enhanced detection capabilities compared to traditional beamforming approaches.

1:45

2pPA4. Realistic terrain boundary conditions for the three-dimensional finite-difference parabolic equation method applied to infrasonic propagation. Codor Khodr, Mahdi Azarpeyvand (Mech. Eng., Univ. of Bristol, 66B Queen's Rd., Bristol, Avon BS8 1QU, United Kingdom, ck15491@bristol.ac.uk), and David Green (AWE Blacknest, Reading, United Kingdom)

Infrasound propagation in the atmosphere can be significantly influenced by environmental parameters such as meteorology and topography. An accurate modelling of these parameters and their effects can improve signal processing of infrasound arrivals and development of more accurate prediction tools. Over the last few decades, much attention has been given to the integration of complex atmospheric features, namely, absorption, turbulence, density, and refraction, into the numerical methods, while topographic effects have usually been neglected. In this respect, the Parabolic Equation (PE) method has been continuously improved to deal with complex configurations encountered in various applications, such as atmospheric

waveguides, ocean acoustics and electromagnetics. We have further extended the Beilis-Tappert (BTPE) generalized coordinate transform to three-dimensional propagation in order to account for out-of-plane scattering by irregular three-dimensional topography, in the presence of air density variation with altitude. The resulting PE method is then implemented into a numerical solver and used to model wave propagation over a set of generic and realistic terrain and atmospheric conditions. A comparison against two-dimensional PE simulations allows us to isolate three-dimensional effects enabled by the out-of-plane propagation and to better understand the underlying physics of atmospheric wave propagation with complex terrain and atmospheric conditions.

2:00

2pPA5. An acoustic source model for the signal produced by ground motion over an underground event in mountainous terrain. Roger M. Waxler and Claus Hetzer (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38677, rwax@olemiss.edu)

A confined, large explosion underground launches an infrasonic signal into the atmosphere generated by the heaving of the ground over the explosion. For example, the last three North Korean nuclear weapon tests produced infrasonic signals detected at great distances form the test site. Under flat terrain, modeling the acoustic source from a confined explosion is a classical problem, similar to that of a baffled loudspeaker. In mountainous regions, the problem is more difficult. A boundary element method has been developed to this end. Given the vertical motion of the ground surface, the model predicts an the acoustic pressure on a spherical section centered over the source. From this, a source function for far field propagation can be extracted. The methods used to produce such a model will be presented.

2:15

2pPA6. Scaling of a gas-combustion infrasound source. Chad M. Smith and Thomas B. Gabrielson (Graduate Program in Acoust., Penn State Univ., PO Box 30, State College, PA 16804, tbg3@psu.edu)

For equal storage pressure, the available energy density from propane-air combustion is substantially higher than the available energy density from expansion of a compressed gas. Ordinary liquid propane is stored at a pressure of about 1 MPa (150 psi). The potential-energy density in air that has been compressed to 1 MPa is about 170 kJ/kg; the combustion-energy density in propane is about 50 000 kJ/kg. In addition, there is a substantial volumetric advantage to propane in that it condenses to a liquid under normal storage pressures. Given the successful demonstration of compressed-gas infrasound sources (Barlett *et al*., J. Acoust. Soc. Am. **141**, 3567, 2017), useful levels of infrasound might also be produced by cycling a burner on and off. Previous work (Smith and Gabrielson, J. Acoust. Soc. Am. **137**, 2407, 2015) with a 90 kW propane burner demonstrated the potential for generating infrasound by periodic thermal expansion of the surrounding air and the levels measured were supported by a simple thermodynamic model for the periodic-heating process. In this paper, we describe attempts to scale these results to higher levels by using a liquid-propane burner from a hot-air balloon.

2:30–2:45 Break

2:45

2pPA7. Characterizing ocean ambient noise using infrasound network. Alexis Le Pichon, Marine De Carlo (CEA, DAM, DIF, -, Arpajon F-91297, France, alexis.le-pichon@cea.fr), Fabrice Ardhuin (IFREMER, LPO/CNRS, Brest, France), Thibault Arnal (CEA, DAM, DIF, Arpajon, France), and Lars Cerana (BGR, B4.3, Hannover, Germany)

The ability of the International Monitoring System (IMS) global infrasound network to detect atmospheric explosions and events of interest strongly depends on station specific ambient noise signatures which include both incoherent wind noise and real coherent infrasonic waves. To characterize the coherent ambient noise, broadband array processing was performed on historical continuous IMS recordings. Ocean wave interactions contribute to the atmospheric coherent ambient noise field, and we apply wave action models to model these microbarom sources. We use two-dimensional energy spectrum ocean wave products to build a global reference database of oceanic noise sources. Then, we compare observed and modeled directional microbarom amplitudes at several stations. The expected benefits of such studies concern the use complementary data to finely characterize the coupling mechanisms at the ocean-atmosphere interface. In return, better knowledge of ambient ocean noise sources opens new perspectives not only by enhancing the characterization of explosive atmospheric events, but also by providing additional integrated constraints on middle atmosphere dynamics and disturbances where data coverage is otherwise sparse.

3:00

2pPA8. Monitoring infrasound from a Tornado in Oklahoma. Brian R. Elbing, Christopher Petrin (Mech. & Aerosp. Eng., Oklahoma State Univ., OSU-MAE, Eng. North 218, Stillwater, OK 74078, elbing@okstate.edu), and Matthew S. Van Den Broeke (Earth & Atmospheric Sci., Univ. of Nebraska, Lincoln, Lincoln, NE)

Tornado-producing storm systems emit infrasound (sound at frequencies below human hearing) up to 2 hours before tornadogenesis. Weak atmospheric attenuation at these frequencies allows them to be detected hundreds of miles away. Hence, passive infrasonic monitoring may be used for long-range study of tornadogenesis as well as characterization of tornado properties if the infrasound can be correlated with flow-field properties. This requires characterization of infrasound during the life of a tornado as well as other background sources. This is being accomplished as part of the Collaboration Leading Operational UAS Development for Meteorology and Atmospheric Physics (CLOUD-MAP) project, a multi-university collaboration focused on the development and implementation of unmanned aerial systems (UAS) and their integration with sensors for atmospheric measurement. The current work focuses on analysis of a small tornado that occurred on May 11, 2017, about 19 km from an infrasonic monitoring station at Oklahoma State University. Results from this tornado will be reported as well as the available radar data.

3:15

2pPA9. Wind noise reduction at infrasound frequencies using large domes. Carrick L. Talmadge (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38655, clt@olemiss.edu)

One ongoing challenge to the use of infrasound for long-range monitoring applications (e.g., remote explosions or natural sources of infrasound such as tornados, volcanos or hurricanes) continues to be the high noise floors experienced at many permanent infrasound stations. International Monitoring System (IMS) infrasound stations typically address this problem using large pipe arrays as wind noise filters. However, these arrays generate significant distortion of higher frequency signals, making them non-ideal signals with significant frequency content above a few Hz. Further, they do not produce sufficient noise reduction during windy (typically daytime) measurements, making their utility generally limited to very quiet wind-noise periods (typically nighttime). Here we report on the development history of a 20-foot dome intended for permanent infrasound installations, and on measurements collected from arrays at the University of Mississippi Biological Field Station and at the Sandia National Laboratory Facility for Acceptance, Calibration, and Testing (FACT) site. The 20-foot dome at the FACT site is collocated with an IMS 18-m rosette filter and a University of Alaska Fairbanks experimental array, and more than a year of data has been collected, allowing for detailed comparisons of the efficacy of these different wind-noise reduction schemes.

3:30

2pPA10. Determining essential scales and associated uncertainties for regional atmospheric infrasound propagation by incorporating three-dimensional weather model forecasts. Ross E. Alter (Cold Regions Res. and Eng. Lab., US 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., US Army Engineer Res. and Development Ctr., Champaign, IL), and D. K. Wilson (Cold Regions Res. and Eng. Lab., US Army Engineer Res. and Development Ctr., Hanover, NH)

Understanding infrasound propagation is important for geophysical and military applications. Infrasound signatures can be detected from larger

sources such as nuclear detonations and from smaller sources such as bridges, dams, and buildings. Infrastructure sources produce signals of lower amplitude, leading to more regional (up to 150 km) propagation. However, current methods for calculating regional infrasound propagation involve assumptions about the atmosphere, such as horizontal homogeneity, that deviate from more realistic environmental conditions and decrease the accuracy of the infrasound predictions. To remedy this issue, we have interfaced three-dimensional forecasts from the Weather Research and Forecasting (WRF) meteorological model with range-dependent parabolic equation propagation models. To test the improvement of infrasound propagation predictions with more realistic weather data, we conducted sensitivity studies with different propagation ranges and horizontal resolutions and compared them to predictions using simplified meteorological parameters. This allowed identification of the scales most ideal for resolving regional infrasound propagation given the limitations of WRF's spatiotemporal resolutions. Additionally, we present results on quantifying uncertainty in these infrasound simulations by using multiple realizations of WRF forecasts to generate a spread of possible outcomes. Finally, we compare the simulated results to experimental data. (Approved for public release; distribution is unlimited.)

3:45–4:10 Panel Discussion

TUESDAY AFTERNOON, 8 MAY 2018 NICOLLET D2, 1:00 P.M. TO 4:55 P.M.

Session 2pPPa

Psychological and Physiological Acoustics and Animal Bioacoustics: Honoring the Contributions of David Kemp to the Discovery of Otoacoustic Emissons and their Utility for Assessing Hearing Function

Glenis R. Long, Cochair
Speech-Language-Hearing Science, Graduate Center CUNY, Graduate Center CUNY, 365 Fifth Ave., New York, NY 10016

Bastian Epp, Cochair
Electrical Engineering, Centre for Hearing and Speech Sciences, Technical University of Denmark, Ørsteds Plads, Building 352, Room 118, Lyngby 2800, Denmark

Chair's Introduction—1:00

Invited Papers

1:05

2pPPa1. Active cochlear mechanics and outer hair cells. Jonathan F. Ashmore (UCL Ear Inst., UCL, UCL Ear Inst., 332 Gray's Inn Rd., London WC1X8EE, United Kingdom, j.ashmore@ucl.ac.uk)

David Kemp's discovery of otoacoustic emissions signalled a new era in cochlear modelling and neurobiology. After Bekesy, the emphasis on cochlear processing had centered on the physics of frequency selectivity with any underlying physiological or cellular contributions taking a back seat. Since then, many lines of evidence have revealed that the (at the time) enigmatic population of outer hair cells (OHCs) within the cochlea plays a central role in modifying the basilar membrane mechanics: OHCs act as fast actuators up to at least 80 kHz in order to cancel fluid damping of the partition. The active mechanical feedback due to OHCs appears to be delicately poised so that small variations from section to section provides a substrate for the inhomogeneities required for OAEs. Metabolically labile, the intrinsic non-linearities of the system can be traced to the hair cell mechano-electrical transduction step itself. Mammalian OHC "motor" function has an identified molecular basis, and only recently have structural studies partially clarified how it this molecule ("prestin"/SLC26A5) works. A further contemporary concern is whether the mammalian cochlea employs the same, but scaled, mechanism of sound amplification for high as well as for low frequencies and this issue will be discussed.

1:25

2pPPa2. Otoacoustic emissions: Laboratory studies and clinical applications. Stephen T. Neely, Daniel M. Rasetshwane, and Michael P. Gorga (Hearing Res., Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, Stephen.Neely@boystown.org)

Historically, the clinical potential of otoacoustic emissions (OAEs) to diagnosis hearing status was recognized prior to any consensus regarding their cochlear origin. In retrospect, we now understand refection-generated emissions as being closely aligned with Kemp's (1980) concept of "cochlear echoes," while distortion-generated emissions are necessarily tied to nonlinearities in cochlear mechanics. Research at the Boys Town National Research Hospital provided evidence supporting the cochlear origin of OAEs by demonstrating an approximate two-times relationship between an estimate of forward-latency derived from auditory brainstem responses and the round-trip latency of OAEs for tone-burst stimuli. Our application of clinical decision theory to distortion-product OAE levels, based on a large

number of both normal-hearing and hearing-impaired ears, helped to establish (1) the validity of using DPOAEs to identify the presence of hearing loss and (2) guidelines for the interpretation of DPOAE levels in the clinic. Our exploration of reflection-generated emissions that we call cochlear reflectance is a continuation of our efforts to further extend the clinical potential of OAEs.

1:45

2pPPa3. The echo that rolled into thunder: Otoacoustic emissions and newborn hearing screening. Beth Prieve (Syracuse Univ., 805 S. Crouse Ave, Syracuse, NY 13244, baprieve@syr.edu)

Kemp's 1978 publication that described "stimulated acoustic emissions" changed the basic understanding of the cochlea and provided a window to assess cochlear mechanics non-invasively in humans. The emissions, referred to early on as "Kemp's echoes," were adopted as the measure of choice by the Rhode Island Hearing Assessment Screening Program, a clinical research study that boldly investigated the ability to screen every baby for hearing loss. This presentation will review how a new, slightly suspicious, physiological measure was used as an integral part of improving hearing healthcare in newborns. The use of distortion-product and transiently evoked otoacoustic emissions as screening tools to identify hearing loss in newborns and their use in the differential diagnosis of hearing loss will be discussed. Clinical findings from infants with and without hearing loss will be presented that support current otoacoustic emission models, and new measures tied to cochlear physiology will be suggested that could lead to improved clinical use. This presentation emphasizes that the science of basic measures from the auditory system has fueled the national and international movement of improving hearing healthcare and habilitation in newborns.

2:05

2pPPa4. David Kemp's contributions to using otoacoustic emissions for protection of noise-exposed ears. Lynne Marshall (U.Conn. & Naval Submarine Med. Res. Lab, 118 Pearl St., Noank, CT 06340-5733, NoankLynneW@gmail.com)

In 1978, David Kemp published a paper that demonstrated a new auditory phenomenon, OAEs. Since then, OAE research expanded broadly, with continued contributions by Kemp. Significantly, Kemp and his colleagues early-on developed commercially-available equipment. Used in the default settings, the equipment was good for hearing screening. But those who wanted to dig deeper could set up solid clinical-research protocols. This allowed translational and clinical research to progress alongside basic research. While OAEs are most often used clinically as pass/fail hearing-screening or site-of-lesion tests, Kemp at the outset inspired other uses, including detection of cochlear changes. He laid out critically important principles that have served as cornerstones for applied OAE research in the area of noise exposure.

2:25

2pPPa5. David Kemp's impact on cochlear modeling and otoacoustic emission measurement and modeling. Carrick L. Talmadge (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38655, clt@olemiss.edu) and Glenis R. Long (Speech-Language-Hearing Sci., Graduate Ctr. CUNY, New York, NY)

Between 1990 and 1998, Drs. Carrick Talmadge, Glenis Long and Arnold Tubis worked together in the field of Hearing Science at Purdue University on a diverse number of topics, including the measurement and modeling of spontaneous otoacoustic emissions, measurement and modeling of stimulus frequency and distortion product otoacoustic emissions (SFOAEs and DPOAEs respectively), and on cochlear modeling. We will explore here linkages between Dr. Kemp's work and our research. This includes his sweep method for measurement of SFOAEs, which lead to the least-squares-fit based method for recording and analyzing SFOAEs and DPOAEs developed by our group. His emphasis on the clinical relevance of otoacoustic emissions implied a linkage between otoacoustic emission data and the underlying function of the cochlea, and this became a main focus of our cochlear modeling efforts. His evidence for two-sources for DPOAEs and his study of the so-called DPOAE "filter shape" were highly influential in the direction of our research at Purdue University.

2:45–3:00 Break

3:00

2pPPa6. Interrelationships among microstructures of otoacoustic emissions and hearing thresholds. James Dewey (Dept. of Otolaryngol. - Head & Neck Surgery, Univ. of Southern California, 1501 San Pablo St., ZNI Rm. 407, Los Angeles, CA 90033, james.dewey@med.usc.edu) and Sumitrajit Dhar (Roxelyn & Richard Pepper Dept. of Commun. Sci. & Disord., Northwestern Univ., Evanston, IL)

Otoacoustic emission (OAE) amplitudes, phases, and delays often exhibit ripples with small changes in stimulus frequency. Similar microstructure patterns are observed in behavioral hearing thresholds and loudness judgements for pure tones. Here, we summarize work demonstrating that there is a common microstructure in hearing thresholds and OAEs elicited by single tones—referred to as stimulus-frequency OAEs (SFOAEs)—when presented at near-threshold levels. The periodicity and strength of this microstructure are related to SFOAE delay and magnitude, respectively. Such relationships are consistent with this microstructure arising from multiple intracochlear reflections of SFOAE energy between its region of generation and the middle ear boundary—an idea proposed by David Kemp in his first reports describing the discovery of emissions from the ear. The details of OAE generation and propagation explain why similar but not identical microstructures are also observed in the ear canal pressure in response to tones (which is the vector sum of the stimulus and SFOAE pressures) and in two-tone evoked distortion-product OAE spectra, as well as why microstructures shift in frequency as the result of cochlear or middle ear manipulations.

3:20

2pPPa7. Temporal dynamics of the generator of stimulated otoacoustic emissions. Sarah Verhulst (Dept. Information Technol., Ghent Univ., Technologiepark 15, Zwijnaarde 9052, Belgium, s.verhulst@ugent.be), Alessandro Altoè (Dept. of Signal Processing and Acoust., Aalto Univ., Aalto, Finland), Stefan Raufer (Speech and Hearing BioSci. and Technol. Program, Harvard Univ., Boston, MA), Karolina Charaziak, and Christopher Shera (Caruso Dept. of Otolaryngol., Univ. of Southern California, Los Angeles, CA)

The early emission work of David Kemp investigated temporal properties of click-evoked otoacoustic emissions (CEOAEs). Suppressor clicks were positioned up to 10 ms before or after the evoking (test) click and reduced the emission amplitude. Intriguingly, suppressors that preceded the test click by 1-2 ms were more effective than simultaneously presented suppressors. This observation seems not to support the hypothesis that emission generation operates on the basis of an instantaneous gain/suppression mechanism, for which simultaneously presented suppressors should be most effective. Kemps' observations thus left the field with an important OAE generation question that inspired several OAE studies. Kemps' results can be explained on the basis of (i) a non-instantaneous gain mechanism at the OAE generation site, or (ii), complex temporal interactions of basilar-membrane impulse responses (BM IRs) that operate with an instantaneous nonlinearity, but yield the observed non-instantaneous suppression properties. We investigated these dynamics using a nonlinear transmission-line model of the human cochlea and found that maximal CEOAE suppression can occur for preceding clicks even when the cochlear nonlinearity is kept time-invariant, supporting hypothesis (ii). Additionally, we used the frequency-dependence of CEOAE suppression to quantify human BM IR duration and cochlear filter tuning, yielding Q_{ERB} estimates of 13.8 F[in kHz]$^{0.22}$.

3:40

2pPPa8. Intracochlear evidence on generation and propagation of distortion product otoacoustic emissions in gerbil cochlea. Wei Dong (ENT, Loma Linda Univ. Health, 11201 Benton St., Loma Linda, CA 92374, wdong@llu.edu)

Otoacoustic emissions (OAEs, Kemp, 1978) are complex signals appearing as a sum of multiple components relating to different cochlear source regions and different active mechanisms thus making it difficult to ascribe them to specific hearing frequencies. Improving the utility of OAEs requires understanding the origin of these multiple components and extracting the information they carry to the ear canal. The current presentation aims to provide direct intracochlear observations on OAE generation and their transmission. Two- or three-tone evoked pressure responses were simultaneously measured in the gerbil ear canal and scala tympani. Distortion product otoacoustic emissions (DPOAEs) and their intracochlear sources as DPs were analyzed. Results demonstrated: 1) DPOAEs come mainly from the f_2 peak region, and extend further basal with increases in primary-tone intensities. These findings are consistent with several notions including: the theoretical prediction of "wave-fixed" or nonlinear generation, and the location of cochlear amplifier; and 2) DPs post-generation may travel both forward to their own best frequency places and in reverse to the stapes. The reverse propagation of DPs was mainly through reverse traveling waves, even though compression waves may also contribute to DPOAE generation, especially, to those produced close to the stapes in the basal cochlear region.

4:00

2pPPa9. Comparative otoacoustic insights. Christopher Bergevin (Phys. & Astronomy, York Univ., 4700 Keele St., Petrie 240, Toronto, ON M3J 1P3, Canada, cberge@yorku.ca)

"Cochlea wave propagation…" These words started off David Kemp's seminal 1978 contribution to auditory biophysics, where he first described "a new auditory phenomenon" [JASA 64(5):1386-1391]. The impact of this report has transcended numerous disciplines, finding an impressive range of scientific and clinical applications (e.g., monitoring cochlear status to ototoxicity, biophysical models of the cochlea, pediatric hearing screening). One area where this is most readily apparent is the "comparative" one: It appears that animals all across the phylogenetic tree exhibit robust OAE activity. Given broad morphological differences (e.g., cochlear length, presence/absence of a flexible basilar membrane and/or stereovillar hair cells), a comparative otoacoustic approach offers remarkable potential for identifying universal biophysical principles at play. This talk will compare/contrast OAE properties (and the associated implications) across a broad range of groups such as: humans, birds, frogs, lizards, ferrets, non-human primates, tigers, and even insects. Furthermore, by examining OAEs in species where the traditional basilar membrane traveling wave is absent (e.g., reptiles), we stand to gain biophysical insight into the precise role of "wave propagation." Taken together, the enormous impact the otoacoustic approach has had towards understanding the biomechanics of the inner ear speaks to the truly profound nature of the "Kempian" discovery.

Contributed Paper

4:20

2pPPa10. The role of the tectorial membrane in cochlear mechanics. Karl Grosh, Amir Nankali, and Aritra Sasmal (Univ. of Michigan, 2350 Hayward St., Dept. of Mech. Eng., Ann Arbor, MI 48109-2125, grosh@umich.edu)

The tectorial membrane (TM) is strategically located in the mammalian cochlea, anchoring the apical pole of the outer hair cell stereocilia and hovering just above the sensory inner hair cell stereocilia. Genetic mutations of TM-specific proteins cause nonsyndromic deafness in humans while experiments in mutant mice clearly demonstrate the impact that manipulations of TM structural proteins have on hearing thresholds and cochlear sensitivity. This direct evidence shows the importance of this cochlear structure. Mathematical models (with Zwsilocki as an early proponent) have long incorporated the TM but have not included a complete electrical and fluidic coupling. We show that an electromechanical model of the active processes in the cochlea combined with a fluid-structure model that strongly couples the basilar membrane and the TM to the cochlear fluids is crucial in understanding the low and high frequency response in a base-to-apex model of the cochlea. Specifically, we show that the onset of nonlinearity at high frequencies is due to a radial resonance of the uncoupled TM and investigate the influence of fluid-loading on the TM and TM viscoelasticity on the low-pass behavior seen in the apex. A connection of this new wave bearing mechanism to otoacoustic emissions will be discussed.

4:35–4:55 Panel Discussion

Session 2pPPb

Psychological and Physiological Acoustics: Binaural Hearing and Psychoacoustic Methodology (Poster Session)

Yonghee Oh, Chair
Otolaryngology-Head and Neck Surgery, Oregon Health & Science University, Mailcode NRC04, 3181 SW Sam Jackson Park Road, Portland, OR 97239

All posters will be on display from 1:00 p.m. to 5:00 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 1:00 p.m. to 3:00 p.m. and authors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

Contributed Papers

2pPPb1. The acoustic images in the fields of knowledge. Peter Francis C. Gossweiler (Instituto de Artes, UFRGS, Trav. Nsa. Sra. de Lourdes, 230 apt. 503 D, Porto Alegre, Rio Grande do Sul 91920-040, Brazil, petergossweiler@gmail.com)

Can we provide different acoustic images, ontologically speaking, from the fields of knowledge? As an engineer that experiences acoustic by processing mathematical calculations (with an evaluative approach), an architect by projecting physical spaces (focusing on audible comfort), an artist by proposing sound enigmas and mind gaps (perhaps proposing an audible discomfort), a musician by dealing with the reflections of his sound making (awaking the pleasure of listening), a religious man by preaching spiritual rituals (addressing a meditative state for his assembled group of people), just to name a few. Thus, our research evokes philosophers such as Martin Heidegger (by the concept of dasein), Michel Certeau (which suggests exercising rebellion in everyday life) and Marie-José Mondzain (with the birth of the homo-spectator, by his desire for images) to point out, with those key operative concepts, outstanding approaches to how we can investigate the internal site specifics of the human being, in his dynamic and unfinished possibilities, the variables of the listening regime on acoustic images in the fields of knowledge.

2pPPb2. Observations on the shape of hearing sensations in pure-tone lateralization. Florian Völk (WindAcoust., Muehlbachstrasse 1, WinDC 86949, Germany, voelk@windacoustics.com), Jörg Encke, Diana Reimann, and Werner Hemmert (Bio-Inspired Information Processing, Tech. Univ. of Munich, Garching, Germany)

In certain conditions, especially with diotic headphone presentation, hearing sensations are located inside the head. Dichotic headphone presentation can result in lateralization: delaying one of the headphone input signals or reducing its amplitude typically pushes the hearing sensation inside the head towards the contralateral ear. Systematic connections between physical stimulus parameters and hearing-sensation properties provide insight into auditory-localization mechanisms and are helpful in designing and evaluating models thereof. While various studies on the lateral displacement of hearing sensations exist, fewer data is available on the respective shape or spatial extent. Based on observations from a pure-tone lateralization experiment, this study aims at taking a first step towards quantifying the typical shape of the corresponding hearing sensations: in two separate experiments, twelve normal-hearing subjects reported a) the overall spatial extent of the hearing sensation or, if applicable, of all simultaneously occurring sensations and b) the number of simultaneously occurring, spatially separated hearing sensations. The data suggests, in line with earlier studies, considerable differences between the results for pure interaural phase and amplitude differences, respectively. Based on the results, an initial empirical description of the stimulus-dependent hearing sensations in pure-tone lateralization summarizes the observations, especially regarding location and spatial shape.

2pPPb3. A real-time method for computing confidence intervals for trial-based psychoacoustic tests. Jakob Hartl (NASA Interns, Fellows & Scholars, Hampton, VA) and Andrew Christian (Structural Acoust. Branch, NASA Langley Res. Ctr., 2 N. Dryden St., M/S 463, Hampton, VA 23681, andrew.christian@nasa.gov)

The vast majority of psychoacoustic testing involves presenting a subject with a series of short, simple questions (trials). A common desire is to track the level of statistical confidence in the results after each trial so that one can stop the test once a threshold of certainty has been surpassed. Typically, this certainty statistic will take the form of a confidence interval (CI). Past methods to compute CIs involve assumptions that may impact the performance of the resultant test-stopping criterion. This talk proposes a method to compute CIs that is free of assumptions regarding: the testing procedure used (e.g., up-down staircase, PEST, non-adaptive), the testing modality used (e.g., n-alternative forced-choice, yes/no), and the model psychometric function used (e.g., logit, Gompertz). The method involves a fixed-quadrature integration of the likelihood function of the model parameters given the data retrieved after each trial. The execution time is demonstrated to be low enough so that the method may run during the inter-trial pause. The method is demonstrated for a 3-alternative forced-choice detection task involving two interleaved transformed up-down staircases (between which the data is pooled) using a scaled normal CDF as the model for the psychometric function.

2pPPb4. Localization of adventitious respiratory sounds. Brian Henry and Thomas J. Royston (BioEng., Univ. of Illinois at Chicago, 851 South Morgan St., MC 063, Chicago, IL 60607, troyston@uic.edu)

In a recent publication [Henry and Royston, J. Acoust. Soc. Am., 142, 1774–1783 (2017)], an algorithm was introduced to calculate the acoustic response to externally introduced and endogenous respiratory sounds within a realistic, patient-specific subglottal airway tree. This work is extended using an efficient numerical boundary element (BE) approach to calculate the resulting radiated sound field from the airway tree into the lung parenchyma taking into account the surrounding chest wall. Within the BE model of the left lung parenchyma, comprised of more than 6,000 triangular surface elements, more than 30,000 monopoles are used to approximate complex airway-originated acoustic sources. The chest wall is modeled as a boundary condition on the parenchymal surface. Several cases were simulated, including a bronchoconstricted lung that had an internal acoustic source introduced in a bronchiole, approximating a wheeze. An acoustic source localization algorithm coupled to the BE model estimated the wheeze source location to within a few millimeters based solely on the acoustic field at the surface. Improved noninvasive means of locating adventitious respiratory sounds may enhance our understanding of acoustic changes correlated to pathology, and potentially provide improved noninvasive tools for the diagnosis of pulmonary diseases that uniquely alter acoustics.

2pPPb5. Binaural perception of stereo noise bursts as a function of burst duration and degree of interchannel coherence. Steven E. Crawford and Mark F. Bocko (Elec. Eng., Univ. of Rochester, 500 Joseph C. Wilson Blvd., Rochester, NY 14627, steven.crawford@rochester.edu)

Amplitude panning for virtual sound source rendering in stereo and multi-channel audio reproduction is well established. Binaural fusion can be represented by vector addition of the acoustic wave propagation vectors from individual speakers; however, this applies only to stereo signals with perfect interchannel coherence and does not provide a description of the perceived sound image when the interchannel coherence is less than complete. Listening tests show that the stereo image formed by continuous incoherent noise fills the space between the loudspeakers; however, for a single short noise burst (a few milliseconds long) the virtual source collapses to a single point in space with its location determined by the position of the maximum of the interchannel correlation function. Concatenation of an ensemble of short noise bursts with a distribution of interchannel correlation function maxima creates a long noise burst displaying a broad peak in its interchannel correlation function; the location and width of which correspond to the apparent location and spread of the virtual sound image. We present a simple model that combines the effective averaging time of the human central binaural fusion mechanism and the interchannel coherence of the stereo signal to predict the spatial properties of virtual sound sources.

2pPPb6. Vision-induced reweighting of binaural localization cues. Maike Ferber, Bernhard Laback (Acoust. Res. Inst., Austrian Acad. of Sci., Wohllebengasse 12-14, Vienna 1040, Austria, mferber@kfs.oeaw.ac.at), and Norbert Kopco (Pavol Jozef Safarik Univ., Kosice, Slovakia)

Normal-hearing listeners apply frequency-dependent weights when combining the two binaural cues (interaural time and level difference, ITD and ILD) to determine the perceived sound source azimuth. Cochlear-implant (CI) listeners, however, rely almost entirely on ILDs. Since current CI systems do not reliably convey ITD information, CI listeners might learn to ignore ITDs and focus on ILDs instead. We investigated whether this reweighting of binaural cues is generally possible. 20 normal-hearing participants, assigned to two groups, completed an experiment in a virtual audio-visual environment. The experiment consisted of a pre-test to establish the initial ITD/ILD weights, a seven-day training, in which visual feedback reinforced one of the cues, and a post-test to measure the effect of training on the weights. Participants' task was to lateralize octave-wide bands of noise (centered at 2.8 kHz) containing various combinations of spatially inconsistent ITD and ILD. In both groups, the lateralization bias related to the reinforced cue declined significantly from pre- to post-test, suggesting that participants reweighted the binaural cues in accordance with the visual feedback. These results are promising in terms of making ITD information usable by CI listeners once it is conveyed by the implants. [Support: SpaCI Danube Partnership project, H2020-MSCA-RISE-2015 #691229.]

2pPPb7. The effect of visual representation of a room on perceived reverberation. Michael Schutte (Div. of Neurobiology, Dept. Biology II, Ludwig-Maximilians-Universität München, Großhaderner Straße 2, Planegg-Martinsried 82152, Germany, michael.schutte@lmu.de), Stephan D. Ewert (Medizinische Physik und Exzellenzcluster Hearing4all, Universität Oldenburg, Oldenburg, Germany), and Lutz Wiegrebe (Div. of Neurobiology, Dept. Biology II, Ludwig-Maximilians-Universität München, Munich, Germany)

In everyday environments, sound reflections from walls and other objects arrive at the listener's ear in addition to the direct sound radiated from a source. Humans have evolved mechanisms to perceptually suppress distracting early reflections, summarized as the precedence effect. Visual information of the surroundings influences the effectiveness of this mechanism. It has also been shown that similar compensation effects can occur for later reflections and reverberation, and that humans can estimate acoustic properties of a room from the visual impression. Taking these findings together, we hypothesize that the visual impression of a room affects the perception of its reverberation. The hypothesis was tested in a highly immersive audio-visual virtual reality environment consisting of a horizontal loudspeaker array in an acoustically treated chamber and a head-mounted display. In a magnitude estimation paradigm, subjects judged the perceived degree of reverberation in conditions where the simultaneously presented acoustic and visual stimuli were either matched regarding the room, sound source azimuth, and sound source distance, or diverged in one of those aspects. Audio-only control conditions served as a baseline. Preliminary results from six subjects suggest a predominant reliance on auditory input when quantifying perceived reverberation even in audio-visually conflicting conditions.

2pPPb8. Toward a visual assistive listening device: Real-time synthesis of a virtual talking face from acoustic speech using deep neural networks. Lele Chen (Dept. of Comput. Sci., Univ. of Rochester, Rochester, NY), Emre Eskimez (Dept. of Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY), Zhiheng Li (Dept. of Comput. Sci., Univ. of Rochester, Rochester, NY), Zhiyao Duan (Dept. of Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY), Chenliang Xu (Dept. of Comput. Sci., Univ. of Rochester, Rochester, NY), and Ross K. Maddox (Departments of 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 a crucial function of the human auditory system, but speech is not only an acoustic signal—visual cues from a talker's face and articulators (lips, teeth, and tongue) carry considerable linguistic information. These cues offer substantial and important improvements to speech comprehension when the acoustic signal suffers degradations like background noise or impaired hearing. However, useful visual cues are not always available, such as when talking on the phone or listening to a podcast. We are developing a system for generating a realistic speaking face from speech audio input. The system uses novel deep neural networks trained on a large audio-visual speech corpus. It is designed to run in real time so that it can be used as an assistive listening device. Previous systems have shown improvements in speech perception only for the most degraded speech. Our design differs notably from earlier ones in that it does not use a language model—instead, it makes a direct transformation from speech audio to face video. This allows the temporal coherence between the acoustic and visual modalities to be preserved, which has been shown to be crucial to cross-modal perceptual binding.

2pPPb9. Effects of vision, listener head movement, and target location on spatial selective auditory attention. Ewan A. Macpherson 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. We have reported [Macpherson & Ellis, ASA Salt Lake, 2016] that listeners' ability to attend to a frontal target in the presence of spatially separated distractors is decreased in a head-movement condition, suggesting that listeners cannot rapidly update the focus of their SSAA to compensate fully for head motion. Participants reported anecdotally that being able to see the target loudspeaker seemed beneficial in directing SSAA. Here, we assessed the benefit of access to a visual reference frame in a similar task. Under lighted and dark conditions, on each trial, listeners either fixated toward 0 azimuth or oscillated their heads at ~0.5 Hz with an amplitude of ~$\pm$40 degrees while five different simultaneous equal-intensity sequences of four spoken digits were presented as target (at azimuths from 0 to 90 degrees) and four distractors (at azimuths $\pm$22.5 and $\pm$45 degrees relative to the central target). Listeners verbally reported the target sequence heard. Performance declined under head motion and with increasing target eccentricity, but not additionally in dark conditions, suggesting that a visual reference frame was not beneficial in directing SSAA in our task.

2pPPb10. Audiovisual speech detection benefits: A comparison across psychophysical approaches. Kaylah Lalonde (Boys Town National Res. Hospital, 555 N. 30th St., Omaha, NE 68131, kaylah.lalonde@boystown.org) and Lynne A. Werner (Speech and Hearing Sci., Univ. of Washington, Seattle, WA)

Visual speech reduces uncertainty as to when auditory speech will occur, thereby enhancing detection of speech in noise. These detection benefits reflect a low-level perceptual mechanism of AV speech enhancement that is independent of the linguistic nature of speech. Audiovisual speech detection benefits have primarily been assessed using two-interval forced-choice paradigms. Results from AV speech detection experiments using three different psychophysical approaches will be presented. These include a single-interval yes/no task, a go/no-go task, and a two-interval forced choice task. In all experiments, adults with normal hearing complete an auditory-only condition with a still image of the talker and an audiovisual condition with a synchronous and congruent visual speech signal. The audiovisual condition includes foil trials/intervals that force participants to use the auditory information to make the decision. Adults demonstrate significant audiovisual benefit to speech detection across all three task, but benefit on the yes/no task depended on the analysis method. Differences in benefit across tasks and signal-to-noise ratios will be discussed. Results of ongoing and planned experiments that aim to isolate the contribution of particular visual temporal cues (onsets, amplitude envelope) will also be presented. [Work supported by NIDCD R01 DC000396, F32 DC015387, and T32 DC005361.]

2pPPb11. Role of expectancy in front-back confusions during sound source localization. William Yost (ASU, PO Box 870102, Tempe, AZ 85287, william.yost@asu.edu)

Front-back confusions (FBCs) often occur in laboratory experiments. Stimulus complexity, especially for high-frequency sounds, and head motion significantly reduce FBCs, so that in the everyday world FBCs probably do not often influence sound source localization. There is a small literature suggesting that listeners "expectancies" (based on experience or other sensory inputs) for where sound sources might be also affect FBCs. A low-pass noise that produces a large number of FBCs and a high-pass noise that does not were used. Listeners indicated the location of sound sources on the azimuth plane. The study aimed to investigate FBCs and not sound source localization accuracy. For one condition, listeners were told that the noises would all be coming from in front, that in another condition from in back, and in a third condition from the full 360 azimuth circle. In the first (front) and second (back) conditions, most of the sounds came from the front or back respectfully, but 20% did not. The expectancy that sounds would be in front or in back had a large influence on FBCs as compared to when sounds were presented from the entire azimuth circle. (Work supported by grants from NIDCD and Oculus VR, LLC.)

2pPPb12. Modeling binaural detection of a Gaussian noise target in the presence of a lead/lag masker. Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu), M. Torben Pastore (Speech and Hearing Sci., Arizona State Univ., Troy, New York), and Yi Zhou (Speech and Hearing Sci., Arizona State Univ., Tempe, AZ)

Recently, we presented the results of a binaural masked detection experiment in which a noise target was temporally embedded within a lead/lag noise masker pair [J. Acoust. Soc. Am. 141, 3639]. The results show that the inter-stimulus interval (ISI) between the masker and its reflection changed the detection threshold significantly. For low ISIs of 2-ms, the average masked detection was -11 dB, but for a greater ISI of 20 ms, the threshold was much higher (-2 dB). In the experiment, masked detection thresholds did not depend on whether the masker lead was on the same side as the target (with the lag on the contralateral side) or the other way round. Different model approaches are presented to simulate the psychophysical results, including a precedence effect (PE) model that was extended using a cepstrum-based method to determine the localization cues for the direct sound of the masker. The PE model is used in conjunction with an EC model and a binaural/monaural detector using optimal frequency-weighting functions that were calculated during the above-threshold conditions during the adaptive experiment. The modeling results show that the PE model is not needed to explain the psychoacoustical results. [Work supported by NSF BCS-1539276 and NSF BCS-1539376.]

2pPPb13. A correlation-based binaural localization model with reflection identification using neural networks. Nikhil Deshpande and Jonas Braasch (Architecture, Rensselaer Polytechnic Inst., 220 3rd St., Troy, NY 12180, deshpn@rpi.edu)

This model identifies a lateralized direct sound and single reflection. Anechoic speech is lateralized using a head-related transfer function. The speech is then time-delayed and lateralized to a different angle. The model then utilizes a binaurally-integrated cross-correlation/auto-correlation mechanism (BICAM) to analyze the lead/lag stimulus and generate a band-limited binaural activity pattern. This output is analyzed to calculate the time delay of the reflection, and the model then uses a neural network to estimate the lateralization of both the direct and reflected sound. From here, the model can remove the reflected copy of the sound, and extract the original raw, anechoic speech. [This work was supported by the HASS Fellowship at Rensselaer Polytechnic Institute, and the National Science Foundation Grant No. NSF BCS-1539276.]

2pPPb14. A comparison of free-field and headphone based sound localization tasks. Devon Brichetto, Brock Carlson, Mary Landis Gaston, Thomas Olson (Neurosci., St. Olaf College, 1520 St. Olaf Ave., Northfield, MN 55057, briche1@stolaf.edu), and Jeremy Loebach (Psych., St. Olaf College, Northfield, MN)

Accurate sound localization is critical for identifying the origin of a sound source and for segregating sources in complex auditory scenes. Sound localization tasks often use free-field presentation of stimuli from a speaker array, or synthetic signals presented over headphones. This study compares localization of ITD and ILD cues presented in the free-field using the SoLoArc (Westerberg *et al*., 2015), a custom designed speaker array that can systematically present auditory and visual stimuli in a 180-degree free-field environment in both the horizontal and vertical planes, with synthetic ITD and ILD cues presented via headphones. Further, we compared two ways for participants to indicate their response: a manual system where participants pointed a laser at the source, which was identified by a spotter, and an automated potentiometer coupled servo system where the voltage derived from the potentiometer is converted into angular location, allowing the participant to indicate source localization without movement of the head or body. Results indicated that localization in the free-field tasks was significantly more accurate than the headphone tasks for both ITD and ILD cues, and that the automated pointing system was superior to the manual system. Pilot data from the elevation task will also be discussed.

2pPPb15. Portable Automated Rapid Testing (PART) measures for auditory research. 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), Aaron Seitz (Psych., Univ. of California, Riverside, Riverside, CA), David A. Eddins (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL), Michelle R. Molis (VA RR&D National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, Portland, OR), Trevor Stavropoulos (Psych., Univ. of California, Riverside, Riverside, CA), Kasey Jakien, Sean Kampel (VA RR&D National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, Portland, OR), Anna C. Diedesch (Commun. Sci. and Disord., Western Washington Univ., Bellingham, WA), Eric Hoover, Karen Bell (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL), Pamela Souza, Melissa Sherman (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL), Lauren Calandruccio, Gretchen Xue (Psychol. Sci., Case Western Univ., Cleveland, OH), Rene Sebena (Psych., Pavol Jozef Safarik Univ., Kosice, Slovakia), and Nirmal Kumar Srinivasan (Audiol., Speech-Lang. Pathol., and Deaf Studies, Towson Univ., Towson, MD)

This presentation will focus on data collected across the globe on tablet computers using applications available for free download, built-in sound hardware, and calibrated consumer-grade headphones. Tests involve spatial release from speech-on-speech masking, binaural sensitivity, gap discrimination, temporal modulation, spectral modulation, and spectrotemporal modulation. The data will be compared across sites and with data from the published literature. The similarity of the obtained data to the expected values and the consistency across sites will help evaluate the potential of this relatively inexpensive and easily disseminated approach to psychoacoustical data collection. The extent to which valid performance can be obtained is a metric of the possibility that in the future such test methods could be used by researchers without access to a full laboratory, clinicians interested in evaluating auditory function beyond the audiogram, and students as part of their training, as well as many other uses not yet imagined.

2pPPb16. A hearing test simulator GUI for clinical testing. Serkan Tokgoz, Yiya Hao, and Issa M. Panahi (Dept. of Elec. Eng., The Univ. of Texas at Dallas, EC33 University of Texas at Dallas 800 West Campbell Rd., Richardson, TX 75080, sxt167830@utdallas.edu)

This paper presents an overview of a useful MATLAB based GUI for hearing testing to evaluate the subjects' hearing ability in noisy environments with different SNR values. With this software package, the examiner will be able to identify which words are correctly perceived and then collect test data in various conditions to measure the performance of the subjects in hearing tests. From the subjects' responses, word recognition rates are saved by the examiner with different noise types such as babble, traffic, machinery, and white noise. Additionally, the subjective tests are completed through repeated testing circles. All hearing testing data and word recognition rate are saved into the database for later use purpose and analysis. This computer aided simulation makes a reliable and cost effective to create real environmental conditions for clinical testing. Our MATLAB based GUI addresses the needs of both clinical evaluation and engineering.

2pPPb17. The effects of interaural level differences on binaural fusion in normal-hearing listeners. Bess Glickman, Yonghee Oh, and Lina Reiss (Otolaryngology-Head and Neck Surgery, Oregon Health and Sci. Univ., 3181 SW Sam Jackson Park Rd., Mail Code: NRC04, Portland, OR 97239, glickman@ohsu.edu)

Sound localization is an important aspect of auditory scene analysis, allowing listeners to group acoustic components from the same location into a single stream (Bregman 1990). Binaural pitch fusion, the fusion of different frequency tones across ears, can be thought of as one type of auditory streaming. Little is known about how sound localization cues affect binaural fusion. The goal of this study was to investigate the effects of interaural level differences (ILDs), one cue for sound localization, on binaural fusion of dichotic tones. Binaural pitch fusion was measured in adult normal-hearing (NH) listeners, using five ILD conditions: ILD = 0, 1.25, 2.5, 5, and 10 dB. Fusion ranges were measured by simultaneous presentation of reference and comparison stimuli in opposite ears, and varying the comparison stimulus to find the frequency range that fused with the reference stimulus. Preliminary results (5 NH; data collection is ongoing) show that small ILDs increase fusion ranges, while larger ILDs decrease fusion ranges. These findings suggest that ILDs can affect fusion range in NH listeners, and imply that simulated sound source location may provide a top-down grouping cue for binaural fusion. [This research was funded by a NIH-NIDCD grant R01 DC013307.]

2pPPb18. Is there an auditory identification "pop-out" for moving speech targets? M. Torben Pastore (Speech and Hearing Sci., Arizona State Univ., 4 Irving Pl., Troy, New York 12180, m.torben.pastore@gmail.com) and William Yost (Speech and Hearing Sci., Arizona State Univ., Tempe, AZ)

Stationary visual targets are often far more salient when the target moves against an otherwise static background—the so-called "pop out" effect. In the first of two experiments, we tested for a similar effect in the auditory domain. Listeners were asked to identify a single word, spoken by a female, in the midst of two or four masking words spoken by males. Listener performance was estimated in terms of percentage of correct responses and compared between conditions where target and maskers were co-located or located at different locations. For some conditions, the target word was amplitude-panned across the loudspeaker array and in others the target remained stationary. Results showed a spatial release from masking for all conditions where the target and maskers were at different locations. The presentation of a stationary versus moving target stimulus yielded no statistically significant difference in identification performance, suggesting that the visual "pop-out" effect may not have a direct corollary in the auditory domain.

2pPPb19. Is there an auditory detection "pop-out" for moving tones in the presence of static noise maskers? M. Torben Pastore (Speech and Hearing Sci., Arizona State Univ., 4 Irving Pl., Troy, New York 12180, m.torben.pastore@gmail.com) and William Yost (Speech and Hearing Sci., Arizona State Univ., Tempe, AZ)

Stationary visual targets often become far more salient when they move against an otherwise static background—the so-called "pop out" effect. In the second of two experiments, we tested for a similar pop-out effect in the auditory domain. Tone-in-noise detection thresholds were measured using a 2-down 1-up adaptive procedure under conditions where the target was stationary or moved via amplitude-panning. Target frequencies of 500 Hz and 4 kHz were tested. Maskers (2–4, depending on the condition) were independent Gaussian noises filtered to have equal energy per octave. All target and masker stimuli were 500-ms duration. Listener performance was compared between conditions where target and maskers were co-located or positioned separately for conditions when the target word was amplitude-panned across the loudspeaker array and in others when the target remained stationary. Results will be presented and discussed.

2pPPb20. Effect of auditory stream segregation cues on binaural pitch fusion. Yonghee Oh, Frederick J. Gallun, and Lina Reiss (Otolaryngology-Head and Neck Surgery, Oregon Health & Sci. Univ., Mailcode NRC04, 3181 SW Sam Jackson Park Rd., Portland, OR 97239, oyo@ohsu.edu)

Binaural pitch fusion, the fusion of dichotically presented tones that evoke different pitches across ears, can be influenced by temporal envelope cues such as coherent amplitude modulation (Oh and Reiss, ASA 2017). This suggests that binaural pitch fusion may be governed at least in part by top-down processes involved in auditory grouping. The current study was designed to investigate how temporal streaming cues either prior or posterior to a fused dichotic tone pair can influence binaural pitch fusion. Six normal-hearing (NH) listeners and six bilateral hearing-aid (HA) users were tested in a modified auditory streaming paradigm (Steiger and Bregman, 1982). A pair of simultaneous, dichotic reference and comparison stimuli was placed in rapid alternation with a third binaural (diotic) capture stimulus, for a total of 10 alternations. The binaural fusion ranges, the frequency ranges over which binaural pitch fusion occurred for dichotic stimuli, were smaller when measured with the streaming paradigm than with static, spectral-only stimuli of equivalent durations. HA users showed greater reductions in fusion ranges with streaming than NH listeners. The findings suggest that temporally flanking stimuli suppress fusion between dichotic stimuli. [Work supported by NIH-NIDCD grant R01 DC013307 and F32 DC016193.]

2pPPb21. Assessing cognitive and perceptual abilities by manipulating task requirements when listening to speech in noise. Rachel E. Miller and Daniel Fogerty (Commun. Sci. and Disord., Univ. of South Carolina, 1229 Marion St, Columbia, SC 29201, remiller@email.sc.edu)

Speech recognition involves both cognitive and perceptual processing that can be differentially affected by the task requirements. This study assessed the degree to which different cognitive and perceptual abilities explained speech recognition using the Coordinate Response Measure (CRM). Speech recognition was assessed by varying the requirements of attention (i.e., selective or divided attention), and by assessing perceptual processes related to speech glimpsing (i.e., steady-state noise versus speech-modulated noise). Additional cognitive and perceptual tasks assessed cognitive interference, working memory, and amplitude modulation detection at 4 and 10 Hz. Results demonstrated that masking release (was most associated with modulation detection, indicating that the ability to take advantage of momentary dips in the masker (i.e., glimpsing) requires better amplitude modulation detection abilities. In contrast, when attentional constraints were imposed, a measure of cognitive interference was most predictive of performance for reporting keywords spoken by both the target talker (i.e., accurate responses) and competitor (i.e., intrusion responses). Interestingly, our measure of working memory was not correlated with any of the CRM tasks. These results suggest that conditions in which the CRM task is administered (i.e. manipulating noise or attention) can differentially weight separate perceptual and cognitive skills. [Work supported, in part, by NIH/NIDCD.]

2pPPb22. Listening to degraded speech can cause listeners to "wait and see". Steven P. Gianakas and Matthew Winn (Speech & Hearing Sci., Univ. of Washington, 1417 NE 42nd St, Seattle, WA 98105, spgia5@uw.edu)

The "wait and see" approach is described as waiting until hearing the end of a sentence or word before committing to a response. Although people with normal hearing (NH) perceive speech in a very rapid incremental fashion, listeners with cochlear implants (CIs) appear to utilize the wait and see approach in tests of word recognition. Specifically, they not only present with lower accuracy, but do not respond (even with eye gaze) until the utterance is fully or nearly complete. This implies that in addition to poorer accuracy scores, CI listeners might be adopting a different, more reluctant approach to word recognition. To better understand why listeners may adopt a reluctant listening strategy, we examined word recognition with a spectrally degraded speech signal in NH listeners, where eye gaze was tracked as a real-time indicator of commitment to a response. Preliminary results show a decrease in accuracy and a delay in response time for NH listeners when the signal is vocoded. However, the delays were not as extreme as those observed in listeners with CIs. These findings would suggest listeners hearing in a severely degraded condition may change their approach to listening, but only after extensive experience adjusting to the signal.

2pPPb23. Differential effects of hearing loss on neural and behavioral measures of speech perception in noise. Tess K. Koerner (VA RR&D National Ctr. for Rehabilitative Auditory Res., 3125 Holmes Ave. South #204, Minneapolis, WI 55408, koern030@umn.edu) and Yang Zhang (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Understanding speech in background noise is difficult for many listeners, and those with hearing impairment tend to show considerable variability in performance. Event-related potentials (ERPs) and associated cortical oscillations are useful for examining neural responses to speech in various listening conditions and may represent neurophysiological markers of auditory perception. The present study examined the effects of hearing impairment on the neural coding and perception of speech in noise in a group of adult listeners with varying degrees of sensorineural hearing loss. This work was also designed to determine whether cortical ERPs and oscillatory rhythms in various frequency bands can predict the effects of hearing impairment on sentence recognition in noise. Passive N1-P2 and MMN responses were elicited with a double-oddball paradigm containing a consonant and vowel change in background noise. Speech perception was evaluated using phoneme discrimination and sentence recognition in noise tasks. Analysis showed that hearing impairment significantly impacted the MMN response but not the obligatory, sensory N1-P2 complex. Results showed that these objective neural responses may be promising neurophysiological markers of perception in noise across listeners. This work has important implications regarding the use of electrophysiological measures for assessing speech perception in complex auditory environments in clinical populations.

2pPPb24. Human recognition of environmental sounds is not always robust to reverberation. James Traer and Josh McDermott (Brain and Cognit. Sci., MIT, 77 Massachusetts Ave., Cambridge, MA 02139, jtraer@mit.edu)

Reverberation is ubiquitous in natural environments, but its effect on the recognition of non-speech sounds is poorly documented. To evaluate human robustness to reverberation, we measured its effect on the recognizability of everyday sounds. Listeners identified a diverse set of recorded environmental sounds (footsteps, animal vocalizations, vehicles moving, hammering, etc.) in an open set recognition task. For each participant, half of the sounds (randomly assigned) were presented in reverberation. We found the effect of reverberation to depend on the typical listening conditions for a sound. Sounds that are typically loud and heard in indoor environments, and which thus should often be accompanied by reverberation, were recognized robustly, with only a small impairment for reverberant conditions. In contrast, sounds that are either typically quiet or typically heard outdoors, for which reverberation should be less pronounced, produced a large recognition decrement in reverberation. These results demonstrate that humans can be remarkably robust to the distortion induced by reverberation, but that this robustness disappears when the reverberation is not consistent with the expected source properties. The results are consistent with the idea that listeners perceptually separate sound sources from reverberation, constrained by the likelihood of source-environment pairings.

Session 2pSA

Structural Acoustics and Vibration and Education in Acoustics: Improving Education in Structural Acoustics and Vibration

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Invited Papers

1:00

2pSA1. Using sports equipment for student exploration of vibration and structural acoustics phenomena. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, drussell@engr.psu.edu)

The familiarity which many undergraduate and graduate students have with sports implements (bats, sticks, clubs, rackets, and balls), renders such objects as suitable subjects for student exploration through laboratory projects and homework problems. This paper will illustrate how problems and projects involving baseball bats, tennis rackets, hockey sticks, and golf clubs can arouse student interests in structural acoustics. Homework problems in which students model the contact time between ball and racket, the frequencies of a golf club shaft, the vibration of the face plate in a golf driver responsible for the trampoline effect, and the radiation of sound from structural modes in hollow and spherical balls provide engaging applications of the theory for membranes, flexural bending in beams, boundary conditions, and radiation. Examples will be shown for using baseball bats, cricket bats, hurling and hockey sticks, ping pong paddles, and tennis rackets in more involved laboratory projects. Sports implements also have the advantage of being non-uniform and asymmetric (wide body and thin handle), non-isotropic (wood grain), or involving complicated geometrics (ellipses) so that students can compare simple theory to more complicated realistic systems.

1:20

2pSA2. $e^{j\omega t}$ and the time domain in Introductory Engineering Acoustics. Karl Grosh (Dept. of Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109-2125, grosh@umich.edu)

Introductory classes in engineering acoustics traditionally focus on frequency domain solutions rather than those in the time domain. While this is somewhat historical in nature, it is also true that extensive coverage of analytic techniques for solving space-time partial differential equations is challenging in one semester. This is not a desirable situation because qualitative and quantitative understanding of the link between the frequency and time domains is often elusive to the new learner of acoustics. Such a deficit stands as a barrier to their deep understanding of the material as well as a knowledge gap of ubiquitous and important techniques. Naturally, computational tools to perform these manipulations are available along with high fidelity digital to analog conversion on students' personal computers. Students now have the capability to play back their simulations to their own ears to compare to experiment. I will present one approach to teaching these concepts to students with no assumed signal processing background using Fourier transforms. I will show how to weave a relatively complicated frequency domain simulation throughout the class and how this model problem can be demonstrated. This same problem also serves to introduce the effects of nonlinear distortion and clipping in transducers.

1:40

2pSA3. 3D-printing and vibration (including resonance). Lawrie Virgin (Dept. of Mech. Eng., Duke Univ., 141 Hudson Hall, Durham, NC 27708-0300, l.virgin@duke.edu)

The material contained in this paper focuses on using 3D printing of relatively simple, flexible structural components and plane frames. The relatively high resolution of modern 3D printers facilitates the production of slender structures, and thus provides an opportunity to exploit geometric parameter variations to enhance a practical understanding of stiffness and vibration. This approach has proved successful in initial inclusion in both the classroom via demonstration models, as well as in the lab in which elementary facilities can be utilized to acquire data. An especially useful facet of this approach is the assessment and justification for modeling simplifications, e.g., judging the circumstances under which the familiar sway assumption is valid in portal frames. Furthermore, minor parametric variations can be easily incorporated into sensitivity studies, and the production of multiple copies of essentially the same geometry promotes an effective use of statistical measurement uncertainty. We then extend this approach to forced vibration, with examples including the vibrating reed tachometer and sympathetic resonance.

2:00

2pSA4. The acoustic guitar as an experimental platform to teach vibration measurements, modal parameter estimation, and data management. Micah R. Shepherd and Tyler P. Dare (Appl. Res. Lab, Penn State Univ., PO Box 30, Mailstop 3220B, State College, PA 16801, mrs30@psu.edu)

The laboratory class required by Penn State's Graduate Program in Acoustics teaches general experimental methods for acoustics. Students learn the basics of data acquisition, sensor calibration and excitation methods through lectures and performing experiments. This talk will discuss the lab experiment used to teach vibration measurements, basic modal parameter estimation and data management using an acoustic guitar as the experimental testbed. Students are required to develop their own test plan for accelerometer and impact hammer locations and then collect the data using a custom-written LabVIEW data acquisition program. Since the data acquisition involves multiple impact and response locations, the students are required to manage their data in a way that allows for easy mapping of data files to spatial locations. They then estimate the guitar body mode shapes by decomposing the measured mobility matrix into its singular values and right- and left-singular matrices. Finally, the students visualize their results by creating a mesh in MATLAB using the impact locations. By completing all of the steps to perform a modal analysis of an acoustic guitar, the students gain experience in both the practical considerations of modal analysis and experimentation in general.

2:20

2pSA5. Demonstrating structural waves in rods/beams. Scott D. Sommerfeldt (Dept of Phys., Brigham Young Univ., N249 ESC, Provo, UT 84602, scott_sommerfeldt@byu.edu)

Structures exhibit some interesting properties in that multiple types of waves can propagate, and the behavior of those waves is different. This demonstration focuses specifically on the difference between longitudinal waves in rods and bending waves in beams. As used here, rods and beams can be the same structure, in practice, but the terms are used to help distinguish between the particular type of wave propagation. For these wave types, the theory of wave propagation for rods and beams is quite accurate in predicting the experimental response, and can be readily demonstrated if near-ideal boundary conditions are used. This demonstration uses a rod/beam that has free-free boundary conditions. For longitudinal waves, the resonances of the rod should be perfectly harmonic. For bending waves, the resonances of the beam are unevenly spaced due to the dispersive nature of the bending waves. It will be shown how the results from a simple experimental apparatus match theory very well, and how the two types of structural waves are consistent with each other. This good agreement between theory and experiment can help reinforce these wave concepts in the minds of the students.

2:40

2pSA6. Case studies in teaching and research mentorship for undergraduates in structural acoustics and vibration. Donald B. Bliss, Mauricio Villa, David Raudales, and Linda P. Franzoni (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., 148B Hudson Hall, Durham, NC 27708, donald.bliss@duke.edu)

A hybrid educational model that mixes classroom learning and individual mentoring is presented for junior and senior undergraduates interested in structural acoustics and vibration. In this activity, talented undergraduates take Duke's first graduate level acoustics course and simultaneously take multiple semesters of independent study for academic credit. The independent studies are co-supervised by faculty members and graduate students. The student research projects are individualized and typically separate from ongoing graduate student research. This structure allows the undergraduate students to feel strong ownership, while giving graduate students teaching and mentoring experience, and providing faculty with a vehicle to explore new ideas. Projects typically have both analytical and experimental components that allow the students to develop their proficiency with visualization and computational tools while developing their physical insight. Several case studies are presented that show impressive results and substantial intellectual growth, including examples of long term commitment to a career in structural acoustics.

3:00 – 3:15

Contributed Papers

3:15

2pSA7. Seismic-infrasound-acoustic-meteorological sensors to dynamically monitor the natural frequencies of concrete dams. Henry Diaz-Alvarez, Luis A. De Jesus-Diaz, Vincent P. Chiarito, Christopher Simpson, and Mihan H. McKenna (GeoTech. Structures Lab., U.S. Army Engineer Res. and Development Ctr., 208 Gridners PL, Vicksburg, MS 39180, henry.diaz-alvarez@usace.army.mil)

The U.S. Army Engineer Research and Development Center (ERDC) is leading research using seismic-infrasound-acoustic-meteorological (SIAM) arrays to determine structural characteristics of critical infrastructure. Large infrastructure, such as dams, emit infrasound (acoustic energy below that of human perception) at their natural modes of vibration, which are related to their structural condition. To validate the concept and its potential use for monitoring flood control structures, a structural evaluation was conducted at the Portuguese Dam in Ponce, Puerto Rico. The dam's dynamic properties were studied prior to the deployment of SIAM arrays using detailed finite element (FE) models assembled in COMSOL Multiphysics software. The principal natural frequencies of the dam were identified and the fundamental modes were confirmed in the infrasound pass-band. The natural frequencies of 4.8 Hz, 6.7 Hz, and 10.2 Hz, respectively, were determined for the lower modes of vibrations. A total of three SIAM arrays were deployed multiple times over the course of three years. This provided a wide range of data to help understand how the dam's natural frequencies changed due to different loading conditions such as the different reservoir levels. Infrasound test results were compared to resonant frequencies calculated from the dam's transient responses produced by ambient and external load excitations.

3:30

2pSA8. Classroom demonstration of acoustic landmine detection using a clamped soil plate oscillator. Jenna M. Cartron (Phys. Dept., U.S. Naval Acad., 3519 Forest Haven Dr., Laurel, MD 20724, Jcartron11@gmail.com) and Murray S. Korman (Phys. Dept., U.S. Naval Acad., Annapolis, MD)

In landmine detection some mines (constructed out of plastic) are difficult to detect using ground penetrating radar. In acoustic landmine detection, one can detect a vibration profile across the surface of the soil (sand) due to the interaction between the soil column and the elastic top plate.

Table top classroom demonstrations can bring to life idealized field experiments. The soil plate oscillator (SPO) apparatus models an idealization of an acoustic landmine detection experiment. The clamped circular elastic plate models the top plate of a plastic buried mine while the soil column supported by the elastic plate models the burial depth. An amplified sinusoidal chirp from a sweep spectrum analyzer drives an AC coil placed below a small magnet attached to the underside of the clamped plate—causing vibration. A laser Doppler vibrometer measured the rms particle velocity at 13 scan locations across the 4.5 inch diameter soil surface in sweeps between 50 Hz and 850 Hz. Results show a fundamental mode shape at 184 Hz (with a central peak) and a second mode at 488 Hz (with peaks to the left and right of the null) corresponding to a second vibration mode.

3:45

2pSA9. Classroom demonstration of nonlinear tuning curve vibration and two-tone tests using a column of glass beads vibrating over a clamped elastic plate. Emily V. Santos and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, santosemily08@gmail.com)

A soil plate oscillator (SPO) apparatus consists of two circular flanges sandwiching and clamping a thin circular elastic plate. The apparatus can model the acoustic landmine detection problem. Here, uniform spherical glass beads—representing a nonlinear mesoscopic elastic material—are supported at the bottom by the acrylic plate (4.5 inch diam, 1/8 inch thick) and stiff cylindrical sidewalls of the upper flange. A magnetic disk centered and fastened below the plate is driven by an AC coil placed below the magnet. Nonlinear tuning curves of the magnet's acceleration are measured by driving the coil with a swept sinusoidal signal applied to a constant current amplifier. Ten (separate) tuning curve experiments are performed using a fixed column of 350 grams of beads using 1,2,3,…,10 mm diameter beads. The backbone curves (peak acceleration vs. corresponding resonant frequency) exhibit a linear region with comparable slopes, while the detailed curvature vs. bead diameter reveals more structure. A bilinear hysteresis model is useful for characterizing results. In two-tone tests, air-borne sound from two 3 inch diameter speakers drives the bead column surface at closely spaced frequencies near the fundamental resonance. Nonlinearly generated combination frequency tones are compared for each of the ten bead diameter experiments.

TUESDAY AFTERNOON, 8 MAY 2018 NICOLLET A, 1:00 P.M. TO 4:00 P.M.

Session 2pSC

Speech Communication: Clinical Populations (Poster Session)

Stephanie A. Borrie, Chair
Department of Communicative Disorders and Deaf Education, Utah State University, 1000 Old Main Hill, Logan, UT 84322

All posters will be on display from 1:00 p.m. to 4:00 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 1:00 p.m. to 2:30 p.m. and authors of even-numbered papers will be at their posters from 2:30 p.m. to 4:00 p.m.

Contributed Papers

2pSC1. The relationship between reading ability and categorical perception of spectrotemporal and spectral contrasts in children. Gabrielle O'Brien, Emily Kubota, and Jason D. Yeatman (Inst. for Learning and Brain Sci., Univ. of Washington, 1417 N.E. 42nd St., Box 354875, Seattle, WA 98105-6246, andronovhopf@gmail.com)

Behavioral evidence suggests that dyslexic readers are impaired in processing rapid temporal changes such as formant transitions. It is unclear if this possible impairment is attributable to the dynamic nature of a formant transition, the brevity of the cue, or a general deficit in categorical decision making. We tested categorical perception of speech in 30 children between the ages of 8 and 12 with a range of reading abilities (dyslexic, below average and above average readers) on two stimulus continua: (1) a /ba/-/da/ continuum, in which a 100-ms formant transition provides a spectrotemporal cue, and (2) a /sa/-/sha/ continuum with a purely spectral contrastive cue also lasting 100 ms. Two test conditions probed the interaction of working memory and categorical perception: a single-interval condition and an ABX paradigm. Children with dyslexia showed shallower psychometric functions and more lapses at the continuum endpoints compared to controls. There was no effect of stimulus continuum or test condition. This suggests that dyslexic children are not specifically impaired in their perception of formant transitions. Rather, dyslexic children may be subtly challenged in auditory coding of, or decision making about, temporally brief cues that may be either spectral or spectrotemporal in nature.

2pSC2. Mandarin vowel and tone identification for Mandarin congenital amusics. Mingshuang Li (Dept. of Commun. Sci. and Disord., Univ. of Texas at Austin, 1 University Station A1100, Austin, TX 78712, limingshuang@utexas.edu), Wei Tang (State Key Lab. of Cognit. Neurosci. and Learning & IDG/McGovern Inst. for Brain Res., Beijing Normal Univ., Beijing, China), Chang Liu (Dept. of Commun. Sci. and Disord., Univ. of Texas at Austin, Austin, TX), Yun Nan, wenjing wang, and qi dong (State Key Lab. of Cognit. Neurosci. and Learning & IDG/McGovern Inst. for Brain Res., Beijing Normal Univ., Beijing, Beijing, China)

Recent studies found that the lexical tone processing in Mandarin Chinese was normal for amusics with musical perception disorders, but was degraded for a subgroup of amusics, named tone agnosics. The current study aimed to further explore whether the lexical deficits in tone agnosics were presented in phonetic processing, in particular, diphthong and triphthong. Vowel and tone identifications were examined among three groups of Mandarin-native listeners: tone agnosics (amusics with lexical tone deficits), the pure amusics (with normal lexical processing) and normal controls. Preliminary results showed that tone agnosics had the lowest scores in the

identification of vowel-plus-tones, vowels, and tones, while the performances were similar between the pure amusics and normal controls. Moreover, tone agnosics showed deficits not only in lexical tone identification, but also in vowel identification. Particularly, the disadvantages of tone agnosics in both vowel and tone processing appeared to be greater in diphthongs and triphthongs than in monothongs. Speech processing of pure amusics was comparable with that of the control group, although they had deficits in musical disorders, implying that musical perception was separate from speech perception at high-level processing. On the other hand, tone agnosia had deficits in both musical and speech processing.

2pSC3. Effects of congenital blindness on the mismatch responses to Mandarin lexical tone, consonant, and vowel. Jie Feng, Xinchun Wu (Dept. of Psych., Beijing Normal Univ., No.19 XinJieKouWai St., Beijing 100875, China, myonly88@126.com), and Chang Liu (Dept. of Commun. Sci. and Disord., Univ. of Texas at Austin, Austin, TX)

With limited visual input, blind individuals rely more on auditory information to perceive the environmental stimuli. In this study, we investigated the effect of congenital blindness on the Mismatch Negativity (MMN) to Mandarin lexical tones, consonants and vowels using the oddball paradigm. The study included 9 congenitally blind subjects and 10 normal-sighted subjects without hearing impairment, while they are matched on age and verbal IQ. For lexical tone processing, no significant difference was observed between the two groups in MMN latencies or amplitudes, while both groups had larger Mandarin tone MMN on the right and central than on the left. For consonant processing, the MMN amplitude was larger on the left and central than the right only for blind participants, whereas MMN latencies of consonants in both groups were shorter on the right than the left. For vowel processing, blind group had longer MMN latencies than control group. These results suggested that blind individuals had similar pre-attentive lexical tone processing, but they relied more on the left and central hemisphere to differentiate consonants, and it took them more time automatically to process different vowels than the sighted individuals, possibly resulting from the compensatory mechanism of neural reorganization in blind subjects.

2pSC4. Monitoring L-Dopa induced dyskinesias (LID) using speech acoustic measures in patients with Parkinson's disease. Emily Wang (Commun. Disord. and Sci., Rush Univ. Medical Ctr., 600 South Paulina St., 1017 Armour, Chicago, IL 60612, emily_wang@rush.edu), Leonard A. Verhagen Metman (Neurological Sci., Rush Univ. Medical Ctr., Chicago, IL), Andong Zhan (Dept. of Comput. Sci., Johns Hopkins Univ., Baltimore, MD), and Carly Blodgett (RML Specialty Hospital in Chicago, Chicago, IL)

More than 90% of patients with Parkinson's disease (PD) treated with levodopa for over 5 years will develop motor fluctuations and involuntary movements known as levodopa-induced dyskinesias (LID). Since LID commonly manifested in the head-neck and trunk areas speech is inevitably affected. In this study, eight acoustic speech measures derived from three speech tasks were selected to examine changes in speech acoustics associated with LID. Twenty subjects with PD, ten during the medication-off state without dyskinesia, and ten during the medication-on state with dyskinesia completed the speech tasks. Results revealed that three speech acoustic measures demonstrated sensitivity to changes associated with LID. In order to capture the changes associated with LID throughout medication cycles, ten PD subjects were provided an Android phone with HopkinsPD, an Android phone application, installed. One-minute speech and video samples were simultaneously collected at three time points over a medication cycle (before, peak, and end dose), twice a day (first morning dose and first afternoon dose) and repeated over three days (every other day) at a patient's home. The results were discussed in light of the long-term goal to develop a reliable at home LID Smart-phone monitoring system that uses speech acoustic measures to monitor LID real-time remotely.

2pSC5. Vowel dynamics in persons with aphasia. Caroline A. Niziolek (Commun. Sci. and Disord., Univ. of Wisconsin–Madison, 1500 Highland Dr., Rm. 485, Madison, WI 53705, cniziolek@wisc.edu) and Swathi Kiran (Speech, Lang. & Hearing Sci., Boston Univ., Boston, MA)

Although aphasia is primarily considered a disorder of language, higher-level grammatical and word-finding deficits are often accompanied by lower-level motoric deficits in the stable production of consonants and vowels. This project investigates the extent to which persons with aphasia (PWA) use feedback to monitor their speech by examining online formant movements during vowel production. PWA and age-matched controls produced 200 repetitions of three monosyllabic words ("eat", "Ed", "add") while neural activity was recorded using magnetoencephalography (MEG). This "speak" condition was interleaved with a "listen" condition in which recorded audio from the speak trials was played back to participants through earphones, serving as a baseline for the neural response. Vowel centering was defined as 2D formant (F1-F2) movement toward the vowel median over the time course of the syllable, lessening acoustic deviation. PWA had greater acoustic variability than controls at vowel onset, but both groups exhibited vowel centering, significantly decreasing variability over the course of the syllable. A hemispheric shift in neural responses to self-produced speech may accompany this corrective movement in PWA. These analyses inform theories of feedforward and feedback influences on speech production in aphasia.

2pSC6. Recovering prosody of a case of foreign accent syndrome. Grace Kuo (Foreign Lang. and Literatures, National Taiwan Univ., 1 Section 4, Roosevelt Rd., Taipei, Taipei City 106, Taiwan, graciakuo@ntu.edu.tw)

Foreign Accent Syndrome (FAS) is a rare disorder characterized by the emergence of a perceived foreign accent following brain damage. In this case study, acoustic analyses were performed on the speech of a Mandarin-speaking female FAS patient at her four doctor's appointments. The reading materials included news in newspaper and a tongue twister. The acoustic analyses include sentence-level intonation and rhythmic measures including %V and PVIs. Results reveal a gradual recovery trajectory from a disfluent stressed-timed pattern to a fluent syllable-timed pattern. A heritage Mandarin speaker and an advanced nonnative speaker recorded the same reading materials. Same acoustic analyses were performed on their speech for comparison.

2pSC7. What does it mean for a voice to be "normal?". Jody E. Kreiman (Head and Neck Surgery, UCLA, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90403, jkreiman@ucla.edu), Anita Auszmann (Linguist, UCLA, Los Angeles, CA), and Bruce R. Gerratt (Head and Neck Surgery, UCLA, Los Angeles, CA)

It is rather unclear what is meant by "normal" voice quality, just as it is unclear what is meant by "voice quality" in general. A clearer understanding of what listeners perceive as normal and what strikes them as disordered would benefit both clinical practice, for which a normal sound is presumably the goal of treatment, and the study of voice quality in general. To shed light on this matter, listeners heard 1-sec sustained vowels produced by 200 speakers (100 male and 100 female), half of whom were recorded in the clinic (ranging from mild to fairly severe pathology) and half of whom were UCLA students with no known vocal disorder. Listeners compared 20 voices at a time in a series of sort-and-rate trials, and ordered them in a line according to the severity of perceived vocal pathology. Any voices perceived as normal were placed in a box at one end of the line. Preliminary results indicate that listeners agreed fairly well about which voices were not normal, but not at all about which were normal. Implications of these findings for evaluation of voice in and out of the clinic will be discussed. [Work supported by NIH and NSF.]

2pSC8. The role of rhythm in adaptation to dysrhythmic speech, continued. Stephanie A. Borrie (Dept. of Communicative Disord. and Deaf Education, Utah State Univ., 1000 Old Main Hill, Logan, UT 84322, stephanie.borrie@usu.edu) and Kaitlin Lansford (School of Commun. Sci. and Disord., Florida State Univ., Tallahassee, FL)

Rhythm cues have been shown to be important for deciphering speech in adverse listening conditions, even when the rhythm cues are corrupted. In an initial attempt to document the relationship between rhythm perception and processing of a naturally dysrhythmic speech signal, we found that listeners with expertise in rhythm perception were not advantaged with initial intelligibility of ataxic dysarthria but were significantly advantaged with adaptation to the degraded speech signal [Borrie, Lansford, and Barrett, *Journal of Speech, Language, and Hearing Research,* **60**, 561–570 (2017)]. We speculated that listeners with skills in rhythm perception are better able to exploit experience (familiarization) with the degraded speech, learning

something useful about corrupted rhythm cues for subsequent processing. This current study investigated whether the relationship between rhythm perception and perceptual adaptation of dysrhythmic speech, observed in our previous study with ataxic dysarthria, holds for other forms of dysrhythmic speech. Here, we replicate our original study with two different forms of dysarthria: the largely irregular and unpredictable speech of *hyperkinetic* dysarthria secondary to Huntington's disease, and the relatively regular and predictable speech of *hypokinetic* dysarthric speech secondary to Parkinson's disease. Results will bear on models of dysrhythmic speech perception as well as clinical practice.

2pSC9. The impact of combined degradations: Intelligibility of disordered speech in noise. Sarah E. Yoho and Stephanie A. Borrie (Communicative Disord. and Deaf Education, Utah State Univ., 1000 Old Main Hill, Logan, UT 84321-6746, sarah.leopold@usu.edu)

In the current study, the intelligibility of dysarthric speech in background noise was determined. Speech-shaped noise was mixed with neurologically healthy (control) and disordered (dysarthric) speech at a series of signal-to-noise ratios. In addition, bandpass filtered control and dysarthric speech conditions were assessed to determine the effect of noise on both naturally and artificially degraded speech. Both the amount of noise and type of speech significantly impacted intelligibility, but there was no interaction between the two factors. Thus, it appears that there is no differential effect of noise on dysarthric speech relative to control speech. Despite this lack of interaction, it is important to note that the intelligibility of dysarthric speech was substantially lower than the intelligibility of control speech at each of the signal-to-noise ratios. This supports the idea that patients with dysarthria and their communication partners should be advised to select a favorable listening environment for communication. Lastly, large-scale online crowdsourcing via Amazon Mechanical Turk was utilized to collect data for the current study. Findings and implications for this data collection approach will be discussed.

2pSC10. Linguistic type-frequency in preschool boys and girls with and without hearing loss. Mark VanDam (Speech & Hearing Sci., Washington State Univ., PO Box 1495, Spokane, WA 99202, mark.vandam@wsu.edu), Jenna Anderst (Commun. Sci. Disord., Eastern Washington Univ., Spokane, WA), Daniel Olds, Allison Saur (Speech & Hearing Sci., Washington State Univ., Spokane, WA), and Paul De Palma (Comput. Sci., Gonzaga Univ., Spokane, WA)

Linguistic type-frequency, how many different lexical types are used, has been examined in usage-based models of child language acquisition. In general, it has been shown that exposure to greater type-frequencies increases children's productive use of language and that language in turn bootstraps later development including language and literacy. It is not currently known if pediatric hearing loss impacts the type-frequency of those children's early communicative productions. In this study, we used a public database available via HomeBank [http://homebank.talkbank.org] to examine the type-frequency in 53 cognitively intact children, 37 with mild to moderate hearing loss (HL) and 16 peers who were typically-developing (TD). For each child, we analyzed 15 minutes of high volubility from a representative daylong recording collected in a natural family setting via an audio recorder worn by the child. Results indicate a main effect of sex favoring girls, but no main effect of HL. There were, however, interaction effects in which TD-boys had a greater type-frequency than HL-boys, but TD-girls were lesser than HL-girls. Within the HL group, HL-girls had greater type-frequency than HH-boys, but within the TD group TD-boys had greater type-frequency than TD-girls.

2pSC11. Audiovisual benefit in the perception of foreign-accented speech by cochlear-implant users. Emily Waddington, Brittany N. Jaekel, Anna R. Tinnemore, Sandra Gordon-Salant, and Matthew Goupell (Hearing and Speech Sci., Univ. of Maryland, College Park, 0100 Lefrak Hall, College Park, MD 20742, emily_waddington@yahoo.com)

When visual and auditory speech cues are presented together, listeners typically obtain an *audiovisual (AV) benefit*, or a boost in speech perception accuracy compared to auditory-only (AO) or visual-only (VO) presentations. Visual cues are particularly important for the speech perception of cochlear-implant (CI) users, who show a greater AV benefit than normal-hearing (NH) listeners. However, whether CI users can utilize visual cues for an AV benefit when the speaker has a foreign accent is unknown. Accented speech can alter auditory (primarily temporal) and visual cues, and is significantly more difficult for CI users in AO listening conditions. We investigated the AV benefit in CI users for unaccented and accented speech, presenting IEEE sentences in VO, AO, and AV conditions. The sentences were spoken by one native English speaker and one non-native English speaker who is a native Spanish-speaker from Peru. Preliminary results reveal that CI users show AV benefits in both unaccented and accented conditions, with substantially larger AV benefits for accented speech. Although visual cues alone may be insufficient for speech perception, CI users may necessitate more reliance on visual cues when presented with auditory cues that have been altered in accented speech.

2pSC12. Processing speed and age predict recognition of spectro-temporally degraded speech. Anna R. Tinnemore (Neurosci. and Cognit. Sci., Univ. of Maryland, College Park, 0100 Samuel J. LeFrak Hall, College Park, MD 20742, annat@terpmail.umd.edu), Lauren Evans (Hearing and Speech Sci., Univ. of Maryland, College Park, Silver Springs, MD), Sandra Gordon-Salant, and Matthew Goupell (Neurosci. and Cognit. Sci., Univ. of Maryland, College Park, College Park, MD)

Cochlear implants (CIs) provide a listener with the temporal envelope of an acoustic signal, but not the spectral fine structure. Speech recognition performance is highly variable among individuals and is often assessed clinically in ideal conditions (over-articulated, clean speech) rather than real-world listening environments. This study examined the effects of age and processing speed on understanding of time-compressed speech by listeners with normal hearing (NH) presented with vocoded speech of varying spectral resolution and listeners with CIs. NH listeners repeated sentences that had been time-compressed to varying degrees (no compression, 20%, 40%, or 60% time compression ratios) and then noise vocoded with various channels (4, 8, or 16 channels or unprocessed), while CI participants repeated unprocessed time-compressed sentences. Performance of age-matched NH listeners on the 4-channel vocoded speech most closely matched CI listeners' performance at each level of time-compression. For NH listeners, age effects were observed on conditions featuring time compression and reduced spectral resolution. Both NH and CI listener groups showed a significant interaction between age and time compression. As difficulty increased (greater time compression for all, fewer channels for NH), cognitive measures of processing speed were correlated with speech recognition performance across all age and listening groups.

2pSC13. How do differences in voice characteristics affect cochlear implant users' speech intelligibility in cocktail-party scenarios? Nawal El Boghdady (Dept. of Otorhinolaryngology, Univ. of Groningen, Univ. Medical Ctr. Groningen, Groningen, Groningen, Netherlands), Etienne Gaudrain (Cognition Auditive et Psychoacoustique, CNRS UMR 5292, Lyon Neurosci. Res. Ctr., Université Lyon 1, Lyon, FR, UMCG, KNO, Huispostcode BB20, PO box 30.001, Groningen 9700 RB, Netherlands, etienne.gaudrain@cnrs.fr), and Deniz Baskent (Dept. of Otorhinolaryngology, Univ. of Groningen, Univ. Medical Ctr. Groningen, Groningen, Netherlands)

Speech intelligibility in the presence of multiple competing talkers — the cocktail-party — is quite challenging for cochlear implant (CI) users. Previous literature has indicated abnormalities in CI users' perception of principal vocal cues (fundamental frequency, F0, and vocal-tract length, VTL), which have been shown to benefit normal hearing (NH) listeners in distinguishing between target and masking speakers in speech-on-speech (SOS) tasks. The hypothesis posited here is that CI users are also expected to benefit from voice differences between target and masking speakers, albeit at a smaller rate compared to NH listeners. Thus, in this study, the benefit in performance in two SOS tasks was measured for both NH (19 participants) and CI users (18 participants) as a function of systematically varying F0 and VTL of the masking talker relative to the target speaker. Results indicate that, contrary to the posited hypothesis and to NH listeners who benefit from differences in F0 and VTL between target and masker, CI users (i) do not benefit from differences in F0, and (ii) demonstrate a slight *decrease* in performance as the difference in VTL is increased. These data have implications on sound coding strategies that should be developed to address these perceptual deficits in CI users. [**Funding:** The University Medical Center Groningen (UMCG), Advanced Bionics, Netherlands Organization for Scientific Research (NWO), VICI Grant No. 91817603.]

2pSC14. Developing finer-grained methods for analyzing word recognition performance by individuals with cochlear implants. Tzu-Ling J. Yu (Speech-Language-Hearing Sci., Univ. of Minnesota, Twin Cities, 1983 Knapp St., Apt. E4, Falcon Heights, MN 55108, yuxxx583@umn.edu), Sarah J. Rosen (Speech-Language-Hearing Sci., Univ. of Minnesota, Twin Cities, St. Paul, MN), Benjamin Munson, and Robert S. Schlauch (Speech-Language-Hearing Sci., Univ. of Minnesota, Twin Cities, Minneapolis, MN)

Cochlear implant (CI) listeners experience overall improvement in speech recognition abilities, but a considerable amount of variability still remains among these individuals. Data from simulated CI listening conditions also show wide variability in monosyllabic word recognition ability. In an adverse simulated condition (2 spectral channels using sinusoidal carriers), the word score can be nearly 0 for some adults with normal hearing. These floor effects pose a challenge for researchers, as they lead to data compression. When analyzing the performance by phonemes, the increase in scores may remove the floor effect. The purpose of this study is to establish the chance performance rate for NU-6 consonant-vowel-consonant (CVC) word lists analyzed by phonemes. To determine this, we randomly paired words from the NU-6 list with CVC words from an on-line corpus of English, the English Lexicon Project, and scored the matches by phonemes using dictionary pronunciations. Thousands of runs of these random pairings were used to generate expected baseline accuracy ranges. These results were compared with individual, behavioral data from people listening to NU-6 words processed using 2-channel sinusoidal vocoders to determine if participants were performing better than chance when scoring by phonemes.

2pSC15. Quality and quantity of infant-directed speech by maternal caregivers predicts later speech-language outcomes in children with cochlear implants. Laura Dilley, Elizabeth Wieland, Matthew Lehet, Meisam K. Arjmandi (Michigan State Univ., Dept. of Communicative Sci. and, East Lansing, MI 48824, ldilley@msu.edu), Derek Houston (The Ohio State Univ., Columbus, OH), and Tonya Bergeson (Butler Univ., Indianapolis, IN)

Caregivers speaking to children often adjust segmental and suprasegmental qualities of their speech relative to adult-directed (AD) speech. The quality and quantity of infant-directed (ID) speech has been shown to support word learning and word segmentation by normal-hearing infants, but the extent to which children with cochlear implants (CIs) benefit linguistically from ID speech is unclear. The present study investigated the extent to which the quantity and quality of ID speech produced in the lab by each of 40 mothers to her child with a CI predicted the child's speech-language outcome measures at two years post-implantation. Multiple measures of ID and AD speech for each mother were taken, including ID speech quantity in one minute, and several measures of ID speech quality, including fundamental frequency characteristics, speech rate, and the area of the vowel triangle formed by corner vowels in F1-F2 space. Forward stepwise regression showed that both quantity and quality of speech significantly predicted language outcomes measured by the Preschool Language Scales, Peabody Picture Vocabulary Test, and the Reynell Developmental Language Scales. These results support the hypothesis that hearing more ID speech that has acoustic modifications typical of IDS promotes language proficiency in children with CIs.

TUESDAY AFTERNOON, 8 MAY 2018 — GREENWAY H/I, 1:00 P.M. TO 5:15 P.M.

Session 2pSP

Signal Processing in Acoustics, Psychological and Physiological Acoustics, Architectural Acoustics, and Noise: Acoustics of Virtual and Augmented Reality

Buye Xu, Cochair
Signal Processing Research, Starkey Hearing Technologies, 6600 Washington Ave. S, Eden Prairie, MN 55344

Jens Meyer, Cochair
mh acoustics, 38 Meade Road, Fairfax, VT 05454

Invited Papers

1:00

2pSP1. Robust data-independent object-related transfer function-based beamformer design for remote listening applications. Hendrik Barfuss, Michael Buerger, and Walter Kellermann (Chair of Multimedia Communications and Signal Processing, Friedrich-Alexander-Universität Erlangen-Nürnberg, Wetterkreuz 15, Erlangen 91058, Germany, hendrik.barfuss@FAU.de)

Virtual reality offers users to experience remote acoustic scenes which are captured in the original environment by a microphone array carried by, e.g., a third person or a robot. To extract and preserve the spatiotemporal nature of specific sounds of interest in the original acoustic environments, data-independent acoustic beamforming appears to be a suitable technique. In this work, we present recent work on robust data-independent beamformer design which is applicable to unconstrained sensor topologies and allows for an intuitive control of the beamformer's robustness. Combined with the concept of polynomial beamforming, flexible beam steering in real

time is possible. Consequently, the beamformer design is well suited for remote listening applications. If the beamformer design is applied to a microphone array, which is integrated into a scatterer, object-related transfer functions (ORTFs) need to be incorporated into the beamformer design. We briefly discuss selected approaches to obtaining the required ORTFs in this work. Finally, we present exemplary results of our robust beamformer design, carried out for a 12-element microphone array integrated into the head of a humanoid robot.

1:20

2pSP2. A planar array of differential microphones for 3D sound capture. Thushara D. Abhayapala, Prasanga N. Samarasinghe, and Hanchi Chen (Res. School of Eng., The Australian National Univ., Canberra, ACT 2601, Australia, thushara.abhayapala@anu.edu.au)

Three dimensional soundfield decomposition based on spherical harmonic analysis is becoming an integral part of 3D audio for virtual and augmented reality systems. Spherical harmonic analysis reveals the underlying characteristics of the soundfield, thus allowing high accuracy manipulation and analysis of the soundfield. This requires a microphone array with 3D pick up capability to detect the soundfield. A well-studied type of such array configuration is the spherical array. Since its geometry coincides with the spherical harmonics, the sound signal captured by a spherical microphone array is well-suited for the spherical harmonic transform. The placement of microphones on a spherical array has to follow a strict rule of orthogonality of the spherical harmonics which limits the flexibility of the array configuration. The spherical shape of the array also pose difficulties on implementation as well as practical usage. Recently, the authors presented the theory and design of a compact hybrid microphone array distributed on a 2D plane for three dimensional soundfield analysis. The proposed array uses a combination of omnidirectional microphones and differential microphones placed along multiple circles on the X-Y plane. In this paper, we present the practical implementation of a prototype array based on the above design.

1:40

2pSP3. Modal beamformer analysis of ambisonics decoders. Jens Meyer (mh Acoust., 38 Meade Rd., Fairfax, VT 05454, jmm@mhacoustics.com), Gary W. Elko (mh Acoust., Summit, NJ), and Steven Backer (mh Acoust., San Francisco, CA)

Ambisonic recording and playback is experiencing a revival with the introduction of higher-order Ambisonics (HOA) and is included in the recent MPEG-H and ATSC 3.0 standards. Higher-Order Ambisonics processing results in an improved spatial resolution compared to the original first-order Ambisonics proposed in 1970s. From a signal processing point of view, Ambisonics is a two-step process. In the first step spherical harmonic signals are generated by an Ambisonic encoder. Based on these spherical harmonic signals an Ambisonic decoder generates the output signals in the second step. Many different decoder flavors have emerged over time. Various decoder designs are motivated by different optimization criterion for the resulting soundfield. In this presentation we will analyze the Ambisonic decoders based on the fact that the two-step Ambisonics processing is closely related to spherical harmonic modal beamforming. It is shown that the decoder output signals can be interpreted in terms of beampatterns. The resulting beampatterns for the different decoders are computed and some performance criteria are presented.

2:00

2pSP4. Directional emphasis in ambisonics. W. Bastiaan Kleijn (Google, Inc., Wellington, New Zealand), Andrew Allen, Jan Skoglund, and Felicia S. Lim (Google, Inc., 345 Spear St., San Francisco, CA, bitllama@google.com)

We describe an ambisonics enhancement method that increases the signal strength in specified directions at a low computational cost. The strengthening method is referred to as a utilization of an enhancement operator. The operator can be used in a static setup to emphasize the signal arriving from a particular direction or set of directions, much like a spotlight amplifies the visibility of objects in one direction. It can also be used in an adaptive arrangement where it sharpens the directionality and reduces the distortions in timbre associated with low-degree ambisonics representations. The enhancement operator can be applied directly to time-domain ambisonics representations but also to time-frequency ambisonics representations.

2:20

2pSP5. A hybrid real-time auralization system for binaural reproduction of virtual acoustic environments with simulated room acoustics adapted for hearing aid users. Florian Pausch, Lukas Aspöck (ITA, RWTH Aachen Univ., Aachen, NRW, Germany), Michael Vorlaender, and Janina Fels (ITA, RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52056, Germany, mvo@akustik.rwth-aachen.de)

A recently developed binaural real-time reproduction system has been extended with an interface to research hearing aids allowing for the conduction of auditory research on subjects with hearing loss. Simulated hearing aid signals based on measurements of generic hearing aid-related transfer functions are additionally processed on a master hearing aid to emulate conventional hearing aid algorithms and played back through the hearing aid receivers. Designed for subjects with mild to moderate hearing loss, the system also facilitates the use of residual hearing capabilities by simulating an external sound field based on generic head-related transfer functions which is reproduced via loudspeakers and acoustic crosstalk cancellation filters. For increased ecological validity, plausible room acoustics are simulated using adjusted simulation models relying on geometrical acoustics. The proposed system was evaluated objectively on different levels by investigating the listening environment and various system components, running a benchmark analysis on the acoustical simulation and auralization, and measuring the combined system latency.

2:40–2:55 Break

2:55

2pSP6. High-resolution room virtualization using first-order source-and-receiver-directional room impulse responses. Franz Zotter, Franck Zagala, Markus Zaunschirm, and Matthias Frank (Inst. of Electron. Music and Acoust., Univ. of Music and Performing Arts, Inffeldfasse 10/3, Graz 8010, Austria, zotter@iem.at)

Virtual acoustic environments benefit from involving the source and receiver directivity to become fully interactive. In this contribution we discuss virtualization of a room based on measured source-and-receiver-directional room impulse responses (SRD RIR), captured by first-order-directional source and receiver arrays. Low order arrays may in some ways overcome a directional versus temporal resolution trade off one encounters in directional measurements. This is done by employing the spatial decomposition method of room impulse responses on both the first-order source and receiver side. We discuss the result of these resolution-enhanced and efficiently measured SRD RIR based on a comparative listening experiment involving the variable-directivity icosahedral loudspeaker (IKO), a spherical beamforming loudspeaker array, whose perceptual sculptural effects have been studied in a room. For the virtualization of the IKO, high resolution directivity measurement of the IKO are inserted, and for its rendering, a high-resolution set of head related impulse responses (HRIRs).

3:15

2pSP7. A raytracing method to create inhomogeneous wave fields for collaborative virtual reality systems. Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu)

A new rapid, raytracing method was developed and implemented for collaborative virtual reality systems. The goal of the method was to auralize historic acoustic venues in Rensselaer's Collaborative-Research Augmented Immersive Virtual Environment Laboratory (CRAIVE-Lab). The CRAIVE-Lab provides a physical/digital environment for collaborative tasks using seamless video from multiple projectors and a 128-channel wave-field system with 6 additional ceiling loudspeakers. For the raytracing method, scanned floor plans are marked up by the user to create a two-dimensional geometric room model to calculate the temporal and spatial early-reflection pattern for the horizontal array of 128 loudspeakers resulting in an individual impulse response for each of the 128 loudspeakers. For this purpose, the loudspeaker array is virtually placed within the floor plan to calculate the impulse responses at each loudspeaker position considering the angles of incidence. The user can define wall materials and a directivity pattern for each sound source. The late reverberation tail is generated using a stochastic model based on the volume and surface characteristics of the space. An individual exponentially decaying reverberation tail is computed for each of the 134 loudspeakers with frequency-specific decay times. [Work supported by NSF Grant Nos. #1229391 and CISL.]

3:35

2pSP8. Achieving realism and repeatability of an orchestra simulated within a concert hall. Matthew T. Neal and Michelle C. Vigeant (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, mtn5048@psu.edu)

Virtual acoustic techniques can allow for direct comparison between concert halls, but care must be taken to ensure the virtual reconstruction connects back to the subtle perceptions of the real environment. To achieve better realism with repeatability between measured halls, a compact spherical loudspeaker array is being used to simulate an orchestra. This sound source is placed at twenty different source locations around the stage. For each instrument measurement position, a swept-sine signal is processed to radiate from the source with a particular instrument's directivity, and the spatial room impulse response (RIR) is measured. This processing is accomplished through filters designed from an instrument directivity database measured with 32 microphones. For each instrument, a set of full-frequency filters, one for each loudspeaker driver, were designed from one-third octave band spherical harmonic weights. These processing techniques are directly analogous to higher order Ambisonics (HOA). In the audience, a 32-element spherical microphone array is used to capture the spatial RIR, and each measurement is convolved with single-instrument anechoic recordings and auralized using HOA. Separate auralizations of each instrument are combined together into a full-orchestral auralization. The measurement setup, filter design, and equalization techniques will be presented. [Work supported by NSF Award 1302741.]

3:55

2pSP9. Perceived stability and localizability in binaural reproduction with sparse HRTF measurement grids. Zamir Ben-Hur (Oculus & Facebook and Ben-Gurion Univ. of the Negev, Be'er Sheva, Be'er Sheva 8410501, Israel, zami@post.bgu.ac.il), David L. Alon (Oculus & Facebook, Ashkelon, Israel), Boaz Rafaely (Elec. & Comput. Eng., Ben-Gurion Univ. of the Negev, Beer Sheva, Israel), and Ravish Mehra (Oculus & Facebook, Redmond, Washington)

With the increased popularity of virtual reality applications, the need for high fidelity spatial audio has emerged. Reproduction of high quality spatial audio requires high resolution individualized head-related transfer functions (HRTFs). However, these are typically unavailable as they demand a large number of measurements and specialized equipment. Given sparse measurements, it is necessary to spatially interpolate the HRTF before employing it in binaural reproduction. Prior studies suggested the use of spherical-harmonics (SH) representation as the basis for the interpolation. However, interpolation of sparse measurements may lead to errors due to spatial aliasing and SH series truncation. In this study, the effects of these errors on the perceived acoustic scene stability and on the localizability of the virtual sound source are investigated numerically, as well as perceptually. Experimental results indicate a significant effect on both attributes due to truncation error, while the effect of the aliasing error is less dominant.

4:15

2pSP10. A computational framework for objective assessment of spatial audio wavefields. Mark F. Bocko and Steven Crawford (ECE Dept., Univ. of Rochester, Rochester, NY 14627, bocko@ece.rochester.edu)

Fusion of binaural acoustic stimuli by humans to form spatial perceptions of virtual sound sources may be described in terms of the cross-coherence properties of a binaurally sampled acoustic wavefield. In this paper, a computational framework for predicting listener perception of acoustic wavefields is presented. A plane-wave expansion of the wavefield, with component amplitudes and phases described by random variables, provides a complete description of the wavefield's spatio-temporal coherence properties. The acoustic source locations and the signals driving the sources, the listening space acoustic response, the location, head orientation, and physiological characteristics of the listener's pinnae, head, and shoulders, and an explicit model of the human auditory system, including the peripheral auditory system and central binaural fusion process, comprise the full model. The source signals serve as inputs to the model and listener percepts indicating the locations and extent in space of the perceived virtual acoustic sources, are the outputs. The framework, which can accommodate constituent sub-models with varying levels of detail and sophistication, provides a useful framework to support the ongoing development of spatial sound rendering systems for applications such as cinema, teleconferencing, and virtual and augmented reality.

4:30

2pSP11. A customizable artificial auditory fovea. Christopher N. Casebeer and Ross K. Snider (Elec. and Comput. Eng., Montana State Univ., Cobleigh 541, Bozeman, MT 59715, christopher.casebee1@msu.montana.edu)

Neuroscience shows that auditory neurons extract a variety of specific signal parameters such as frequency, amplitude modulation, frequency modulation, and onsets. Furthermore, biology gives us examples of auditory foveae in bats and owls where there is overrepresentation of behaviorally relevant signal parameters. This shows that biology finds nuances of these parameters to be more important than compact coding considerations such as orthonormal basis functions. In contrast to typical spectrogram approaches where a specific time window is chosen and where the Fast Fourier Transform provides non-overlapping orthogonal rectangular tilings, we start with the smallest tiling possible. The uncertainty principle gives us this tiling as the Gaussian modulated sinusoid. We can rotate this tiling in the time-frequency plane, which gives us a chirplet. We then construct many chirplets (a chirplet set) centered at the same time-frequency point. We then find the chirplet that best matches the signal at this point. Our ability to sample at arbitrary locations in the time-frequency plane with different chirplet sets gives us the ability to create customizable artificial auditory foveae. We use this chirplet front-end for a classification task of identifying Marmoset vocalizations and comparisons to typical spectrogram methods for audio classification are made.

4:45

2pSP12. Automatic recognition and immersive representation of environmental soundscapes. Mallory M. Morgan and Jonas Braasch (Rensselaer Polytechnic Inst., Greene Bldg, RPI, Troy, NY 12180, morgam11@rpi.edu)

The goal of this research is to automatically analyze a spatial auditory scene using a 16-channel spherical microphone array and produce an automatic transcript of events that occurred in different directions. The system can be used to establish an automatic protocol for collaborative business meetings or to monitor natural environments. This paper focuses on the automatic data collection of environmental and animal sounds in conjunction with the RPI/IBM Jefferson Project to sensor Lake George, NY. The amount of audio data required for bioacoustics monitoring and other applications is often too large to manually sort the data. Consequently, automatic environmental sound recognition techniques have to be applied to automatically (1) identify and extract relevant acoustic stimuli and (2) classify these stimuli after a training period. Using spatial audio data collected over days in forested areas in Upstate New York, information is automatically extracted and analyzed to find the most relevant parts of these large data sets. The analysis can then be used to create an automated non-linear time-lapse to summarize the events that occurred and meaningfully re-represent them in our immersive virtual environment, CRAIVE-Lab —together with the collected visual material. [Work supported by NSF No. 1631674 and CISL.]

5:00

2pSP13. Multiple sound-source localization and identification with a spherical microphone array and lavalier microphone data. Jonathan Mathews and Jonas Braasch (Architectural Sci. - Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St, Greene Bldg., Troy, NY 12180, mathej4@rpi.edu)

Spherical microphone arrays have attained a considerable amount of interest for their dynamic beamforming and tracking capabilities. Although many robust algorithms have been developed to accurately localize individual sources, multiple source localization, especially for concurrent or colocated sources, has proved to be a challenging topic. This project demonstrates a technique to isolate individual concurrent sources by using lavalier microphone data to determine the relative difference in signal energy between sources. This data is then incorporated into the source-tracking algorithm governing the beamforming spherical array, a particle filter, by allowing the rejection of frames containing concurrent source information. This improves the tracking ability of the system. [Work supported by Rensselaer HASS Fellowship, NSF No. 1631674, and CISL.]

TUESDAY AFTERNOON, 8 MAY 2018 MINNEHAHA, 2:00 P.M. TO 3:00 P.M.

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 Drive 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 that those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 8 May 2018.

Scope of S3/SC 1: Standards, specifications, methods of measurement and test, instrumentation and terminology in the field of psychological and physiological acoustics, including aspects of general acoustics, which pertain to biological safety, tolerance, and comfort of non-human animals, including both risk to individual animals and to the long-term viability of populations. Animals to be covered may potentially include commercially grown food animals; animals harvested for food in the wild; pets; laboratory animals; exotic species in zoos, oceanaria, or aquariums; or free-ranging wild animals.

TUESDAY AFTERNOON, 8 MAY 2018 MINNEHAHA, 3:15 A.M. TO 4:30 P.M.

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

P. B. Nelson, Vice Chair ASC S3
Department of SLHS, University of Minnesota, 115 Shevlin, 164 Pilsbury Drive S.E., Minneapolis, MN 55455

Accredited Standards Committee S3 on Bioacoustics. Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note that those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 8 May 2018.

Scope of S3: Standards, specifications, methods of measurement and test, and terminology in the fields of psychological and physiological acoustics, including aspects of general acoustics which pertain to biological safety, tolerance, and comfort.

Meeting of Accredited Standards Committee (ASC) S1 Acoustics

R. J. Peppin, Chair ASC S1
5012 Macon Road, 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 Road
Aberdeen Proving Ground, MD 21005 5425

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

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics, ISO/TC 43/SC 3, Underwater acoustics, and IEC/TC 29 Electroacoustics, take note that those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 8 May 2018.

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

TUESDAY EVENING, 8 MAY 2018 7:30 P.M. TO 9:00 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings. All meetings will begin at 7:30 p.m., except for Engineering Acoustics which will hold its meeting starting at 4:30 p.m.

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

Committees meeting on Tuesday are as follows:

Engineering Acoustics (4:30 p.m.)	Greenway D
Acoustical Oceanography	Greenway A
Animal Bioacoustics	Lakeshore B
Architectural Acoustics	Nicollet D3
Musical Acoustics	Greenway F/G
Physical Acoustics	Greenway J
Psychological and Physiological Acoustics	Greenway D2
Structural Acoustics and Vibration	Greenway C

Session 3aAA

Architectural Acoustics, Psychological and Physiological Acoustics, and Speech Communication: Auditory Perception in Virtual, Mixed, and Augmented Environments

Philip W. Robinson, Cochair
Media Technology, Aalto University, PL 15500, Aalto 00076, Finland

G. Christopher Stecker, Cochair
Hearing and Speech Sciences, Vanderbilt University, 1215 21st Ave. South, Room 8310, Nashville, TN 37232

Chair's Introduction—8:50

Invited Papers

8:55

3aAA1. Validating auditory spatial awareness with virtual reality and vice-versa. G. Christopher Stecker, Steven Carter, Travis M. Moore, and Monica L. Folkerts (Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232, g.christopher.stecker@vanderbilt.edu)

"Immersive" technologies such as virtual (VR) and augmented reality (AR) stand to transform sensory and perceptual science, clinical assessments, and habilitation of spatial awareness. This talk explores some of the general challenges and opportunities for VR- and AR-enabled research, illustrated by specific studies in the area of spatial hearing. In one study, freefield localization and discrimination measures were compared across conditions which used VR to show, hide, or alter the visible locations of loudspeakers from trial to trial. The approach is well suited to understanding potential biases and cuing effects in real-world settings. A second study used headphone presentation to understand contextual effects on the relative weighting of binaural timing and intensity cues. Previous studies have used adjustment and lateralization to "trade" time and level cues in a sound-booth setting, but none have attempted to measure how listeners weight cues in realistic multisensory scenes or in realistic temporal contexts. Finally, a third study used simple VR "games" to measure implicit awareness of reverberation in multi-target scenes. Listeners' tolerance of reverberant mismatch can be used to assess and habilitate auditory spatial awareness and provides a testbed for future applications of auditory augmented reality.

9:15

3aAA2. Designing rehabilitative experiences for virtual, mixed, and augmented reality environments. 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), Aaron Seitz (Psych., Univ. of California, Riverside, Riverside, CA), Timothy J. Vallier, and Dawna Lewis (Boys Town National Res. Hospital, Omaha, NE)

It is rapidly becoming possible to conduct nearly all of the diagnostic testing necessary for understanding suprathreshold auditory and visual processing abilities using consumer-grade electronics. It is also the case that all of the psychometric methods used are consistent with a game-based structure, which will encourage engagement and increase reliability of the testing. The obvious next step is to integrate diagnosis and rehabilitation into a single game-based framework, for which virtual, augmented, and mixed reality are an ideal platform. The possibilities include auditory, visual, and memory training and diagnostic games, as well as guided interactions to improve skills in real-life scenarios. Essential to effective design, however, is optimizing interfaces to facilitate universal access. Universal design is essential for rehabilitation because it is necessarily the case that those in need of services cannot be assumed to have optimal auditory, visual, tactile, and balance capabilities. The most exciting challenge for the future will involve designing environments that scale appropriately to engage, challenge, and improve all of the senses and cognitive systems of the participant while being flexible enough to provide engaging experiences for those who can most benefit from the experience.

Contributed Papers

9:35

3aAA3. Effects of non-individual head-related transfer functions and visual mismatch on speech recognition in virtual acoustic environments. Kristi M. Ward (Northwestern Univ., 2240 N. Campus Dr., Rm. 2-381, Evanston, IL 60208, kmward@u.northwestern.edu), Z. Ellen Peng (Univ. of Wisconsin–Madison, Madison, WI), and Tina M. Grieco-Calub (Northwestern Univ., Evanston, IL)

There is widespread research and clinical interest in quantifying how the acoustics of real-world environments, such as background noise and reverberation, impede a listener's ability to recognize speech. Conventional methods used to quantify these effects include dichotic listening via headphones in sound-attenuated booths or loudspeakers in anechoic or low-reverberant environments, which lack the capability of manipulating room acoustics. Using a state-of-the-art Variable Room Acoustics System housed in a virtual sound room (ViSoR), this study aims to systematically assess the effects of non-individual head-related transfer functions (HRTFs) and mismatched visual perception on speech recognition in virtual acoustic environments. Young adults listened to and repeated sentences presented amidst a co-located two-talker speech competitor with reverberation times ranging from 0.4 to 1.25 s. Sentences were presented in three listening conditions: through a loudspeaker array in ViSoR with the participants' own HRTFs (*Condition 1*); via headphones in a sound-attenuated booth with non-individual HRTFs(*Condition 2*); and using the same binaural reproduction as *Condition 2* in ViSoR (*Condition 3*). *Condition 3* serves as a control condition, allowing us to quantify the separate effects of non-individual HRTFs and visual mismatch on speech recognition. Discussion will address the validity and use of virtual acoustics in research and clinical settings.

9:50

3aAA4. Use of non-individualized head-related transfer functions to measure spatial release from masking in children with normal hearing. Z. Ellen Peng, Ruth White, Sara Misurelli, Keng Moua, Alan Kan, and Ruth Litovsky (Waisman Ctr., Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53711, zpeng49@wisc.edu)

Spatial hearing studies with children have typically been conducted using loudspeakers in laboratories. However, loudspeaker arrays are rare in clinics due to high cost and technical set-up requirements. The use of virtual auditory space (VAS) with non-individualized head-related transfer functions (HRTFs) can increase the feasibility of assessing spatial hearing abilities in clinical settings. A novel paradigm for measuring spatial release from masking (SRM) was developed using non-individualized HRTFs. This paradigm measures the minimum angular separation needed between target and masker to achieve a 20% increase in target speech intelligibility. First, the 50% speech reception threshold (SRT) was measured with target and masker co-located to one side. Then, the masker position was adaptively changed to achieve 70.7% intelligibility while maintaining the signal-to-noise ratio at the level of the co-located SRT. To verify the use of non-individualized HRTFs, normal-hearing children were tested (1) using a loudspeaker array and (2) in headphone-based VAS created using KEMAR HRTFs measured in the same setup as (1). Preliminary results showed that co-located SRTs and target-masker angle separation to achieve a 20% SRM were similar in loudspeaker array and in headphone-based VAS. This suggests that non-individualized HRTFs might be used in an SRM task for clinical testing.

10:05

3aAA5. How physical versus panned sources in dry or reverberant conditions affect accuracy of localization in sound field synthesis systems. Anna C. Catton, Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, anna.catton@huskers.unl.edu), Adam K. Bosen, Timothy J. Vallier, and Douglas H. Keefe (Boys Town National Res. Hospital, Omaha, NE)

Sound field synthesis systems vary in number and arrangement of loudspeakers and methods used to generate virtual sound environments to study human hearing perception. While previous work has evaluated the accuracy with which these systems physically reproduce room acoustic conditions, less is known on assessing subjective perception of those conditions, such as how well such systems preserve source localization. This work quantifies the accuracy and precision of perceived localization from a multi-channel sound field synthesis system at Boys Town National Research Hospital, which used 24 physical loudspeakers and vector-based amplitude panning to generate sound fields. Short bursts of broadband speech-shaped noise were presented from source locations (either coinciding with a physical loudspeaker location, or panned between loudspeakers) under free-field and modeled reverberant-room conditions. Listeners used a HTC Vive remote laser tracking system to point to the perceived source location. Results on how physical versus panned sources in dry or reverberant conditions impact accuracy and precision of localization are presented. Similar validation tests are recommended for sound field synthesis systems at other labs that are being used to create virtual sound environments for subjective testing. [Work supported by NIH GM109023.]

10:20–10:35 Break

10:35

3aAA6. Evaluating immersion of spatial audio systems for virtual reality. Lukas Aspöck, Michael Kohnen, and Michael Vorlaender (ITA, RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52056, Germany, mvo@akustik.rwth-aachen.de)

Multiple aspects influence the quality of experience of different VR presentations. One popular aspect which is usually considered for rating the presentation is the concept of immersion. Its complex nature as well as indistinct definitions make it challenging to use it as an objective measure in scientific experiments. Additionally previous studies revealed contradictory definitions of immersion and its separation from the concept of presence. To investigate immersion, a nomological net was developed which connects various items contributing to immersion. These items were assigned to subcategories and should ideally be well defined and measurable. For each item, multiple questions were formulated. Pre-testing on their linguistic quality and unambiguity was conducted to identify suitable questions for each item. These questions were applied in two listening experiments: A between-group design for the evaluation of the chosen questions and a within-subject design for the evaluation of differences between Higher-Order Ambisonics, VBAP, and binaural loudspeaker reproduction.

10:50

3aAA7. Perceptually plausible room acoustics simulation including diffuse reflections. Oliver Buttler (Medizinische Physik and Cluster of Excellence Hearing4all, Universität Oldenburg, Oldenburg, Germany), Torben Wendt, Steven van de Par (Acoust. Group and Medizinische Physik and Cluster of Excellence Hearing4all, Universität Oldenburg, Oldenburg, Germany), and Stephan D. Ewert (Medical Phys. and Cluster of Excellence Hearing4all, Universität Oldenburg, Carl-von-Ossietzky Str. 9-11, Oldenburg 26129, Germany, Stephan.ewert@uni-oldenburg.de)

Immersive and convincing acoustics in virtual reality applications require computationally efficient methods. The fast and perceptually plausible room acoustics simulator [RAZR, see Wendt *et al.*, JAES, 62, 11 (2014)] approaches this demand by drastic simplifications with respect to physical accuracy while still accomplishing perceptual plausibility. RAZR is based on a hybrid approach where early reflections are calculated as image sources for a shoebox-room approximation up to a low order, and the later reverberation is generated by a binaurally extended feedback-delay-network (FDN). Although it was demonstrated that a good perceptual agreement with real non-shoebox rooms is achieved, the shoebox-room simplification might cause limitations for rooms which strongly diverge from this assumption. Here the perception of temporal smearing of early diffuse reflections, effectively simulating effects of scattering and multiple reflections caused by geometric disturbances at walls and by objects in the room, was systematically assessed. A single parameter was introduced to quantify

3a WED. AM

deviations from an empty shoebox room. It was demonstrated that perceptually plausible results can be obtained for auralized natural stimuli and for the binaural impulse responses themselves. It is shown how parameters in RAZR are derived from room geometry and surface materials or from measured BRIRs or frequency-dependent reverberation times.

11:05

3aAA8. Evaluation of near field distance perception in virtual environments. Philip W. Robinson (Oculus Res., PL 15500, Aalto 00076, Finland, philrob22@gmail.com)

Binaural synthesis of spatial audio for virtual and augmented environments is typically performed by convolving an anechoic signal with impulse responses measured at the listener's ears using a source at a fixed distance. This method is accurate for sounds presented at the measured impulse response's distance, and for more distant sources, decreasing intensity provides a reasonable approximation. However, sources nearer than 1 m provide additional distance cues. The head-shadow effect becomes markedly stronger and exponential sound intensity falloff comes into play, exaggerating inter-aural level differences. Also, the angle to each ear diverges, changing the directionally dependent spectral filtering of the pinnae. Changes in inter-aural level differences can be approximated across listeners, but changes in pinna cues with changes in distance are highly individual, and thus less easily approximated. Reproducing some or all of these cues for each individual may be necessary to create a convincing percept of very near objects in virtual and augmented reality environments. The present work aims to determine the importance of each cue for static and dynamic sources. Listening tests have been conducted in a virtual environment using generic and individual head related impulse responses, with and without near-field compensation.

Invited Papers

11:20

3aAA9. Improving the perception of a sound source's polar angle in mixed reality. Hannes Gamper and Ivan J. Tashev (Res., Microsoft, One Microsoft Way, Redmond, WA 98052, hannes.gamper@microsoft.com)

Mixed reality applications blend real and virtual scenes. To render virtual objects in a scene, the rendering system needs to accurately control their perceived location. In the acoustic domain, the location of a sound source is often given in interaural coordinates, i.e., as the lateral and polar angle and distance relative to the midpoint of the listener's interaural axis. This description is useful as it allows separating the effect of various perceptual cues on each interaural spatial dimension. Prior research has shown that the human perception of a sound source's polar angle, i.e., the angle of rotation about the interaural axis, tends to be less accurate than the perception of its lateral angle, i.e., the angle off the median plane. When rendering virtual sound sources, listeners often confuse locations above and below the horizontal plane or in the front and in the back. Here, we review cues that affect the perception of polar angle as well as approaches to improve the accuracy of rendering the polar angle of a virtual sound source in mixed reality applications.

11:40

3aAA10. Localization and externalization in binaural reproduction with sparse HRTF measurement grids. Zamir Ben-Hur (Oculus & Facebook and Ben-Gurion Univ. of the Negev, Be'er Sheva, Israel), David L. Alon (Oculus & Facebook, Menlo Park, CA), Boaz Rafaely (Elec. and Comput. Eng., Ben-Gurion Univ. of the Negev, Beer Sheva, Israel), and Ravish Mehra (Oculus & Facebook, 8747 148th Ave. NE, Redmond, WA 98052, ravish.mehra@oculus.com)

High-quality spatial sound reproduction is important for many applications of virtual and augmented reality. A key component in spatial audio reproduction is the head-related transfer function (HRTF). To achieve a realistic spatial audio experience, in terms of sound localization and externalization, high resolution individualized HRTFs are necessary. However, these are typically unavailable as they require specialized equipment and a large number of measurements, which motivates the use of sparsely measured HRTFs. Reducing the number of measurement points requires spatial interpolation, and may lead to spatial aliasing error. Previous studies suggested the use of spherical-harmonics (SH) decomposition for the interpolation. With a sparse grid, the SH representation is order-limited, leading to a constrained spatial resolution due to the truncation error. In this study, the effect of sparse measurement grids on the reproduced binaural signal is perceptually evaluated, through localization and externalization tests under varying conditions of aliasing and truncation. Preliminary results indicate relatively large effect of SH order on these attributes, while smaller effect is observed due to aliasing error.

Session 3aAO

Acoustical Oceanography, Animal Bioacoustics, and Underwater Acoustics: Ambient Noise Oceanography in Polar Regions: Noise Properties and Parameter Estimation

Oskar Glowacki, Cochair
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Grant B. Deane, Cochair
Marine Physical Lab., Univ. of California, San Diego, 13003 Slack St, La Jolla, CA 92093-0238

Chair's Introduction—8:15

Contributed Paper

8:20

3aAO1. Characteristics of the under-ice soundscape in the southern Beaufort Sea during Ice Exercise 2016. John E. Joseph, D. Benjamin Reeder, and Tetyana Margolina (Oceanogr., Naval Postgrad. School, 833 Dyer Rd, Monterey, CA 93943, jejoseph@nps.edu)

Ice Camp SARGO was the remote hub of operations for the multinational naval operation Ice Exercise 2016 (ICEX-16), held in March of that year. Over a three-day period in early March, continuous recordings of the under-ice soundscape were collected with receivers deployed at various depths through first-year ice in the vicinity of the ice camp as it drifted westward across the Beaufort Sea approximately 175 nm north of Prudhoe Bay, Alaska. A significant reduction in the strength of easterly winds resulted in deceleration of the ice sheet during the period of observation, inducing notable ice cracking and ridging events near the camp. Ice sheet movement slowed from about 0.5 knots early in the test to virtually coming to a halt near the end of the recording period. Sounds from naturally occurring and anthropogenic sources in the 10-Hz to 10-kHz band detected in the recordings were analyzed in connection to the origins of the sound and correlated to varying environment conditions including wind speed and ice motion. Results show a wide variety of persistent and transient sound sources contribute to the total soundscape at this remote location. Comparisons to other soundscape observations in the region are discussed.

Invited Paper

8:35

3aAO2. Underwater noise measurements during a year long Shallow Water Canada Basin Acoustic Propagation experiment from 2016 to 2017. Mohsen Badiey (Univ. of Delaware, Newark, DE 19716, badiey@udel.edu), Ying-Tsong Lin (Woods Hole Oceanographic Inst., Woods Hole, MA), Sean Pecknold (DRDC Atlantic Res. Centere, Dartmouth, NS, Canada), Altan Turgut (Naval Res. Lab., Washington, DC), Megan S. Ballard, Jason D. Sagers (Appl. Res. Labs., Austin, TX), and Christopher Whitt (JASCO Appl. Sci., Dartmouth, NS, Canada)

A year-long, multi-institution, acoustical oceanographic measurement on the Chukchi Continental Shelf region of the Canada Basin started on October 2016 is reported. Ten vertical receiver line arrays and a horizontal receiver line array were deployed together with oceanographic sensors to measure the sound speed profiles, ice formation, and currents over an area of approximately 30 km by 50 km. Various aspects of the noise including spectral fluctuations, directionality, seasonal dependence, and intensity fluctuations are studied over time. In this paper, we present preliminary measurement results depicting spatial and temporal distribution of underwater background noise at the experiment site. [Work supported by ONR321 OA.]

Contributed Papers

8:55

3aAO3. Shallow water propagation of mid-frequency cross-shelf acoustics on the Chukchi Shelf. Justin Eickmeier, Mohsen Badiey (Univ. of Delaware, 17 McCord Dr., Newark, DE 19713, jeickmei@udel.edu), and Altan Turgut (Acoust. Div., Naval Res. Lab, Washington, DC)

The shallow water Canada Basin Acoustic Propagation Experiment (SW CANAPE) was conducted to study the effects of oceanographic variability on broadband acoustic fields in the Arctic. The physics of the acoustic waveguide on the northeastern edge of the Chukchi Shelf are influenced by dynamic boundary conditions and spatio-temporal fluctuations in the water column temperature and salinity profiles. Several oceanographic and acoustic receiving arrays were deployed across the Chukchi Shelf out to the shelf break region. Linear frequency modulated (LFM) signals were transmitted by two sources on the shelf for a long period of time. The influence of small scale, short-term water column variability, and dynamic upper boundary conditions including open water, marginal, and solid ice zones on shallow water propagation is shown for a 10 km source-receiver separation with well-defined water column properties measured at the source, receiver, and a mid-point along the cross-shelf acoustic path. [Work supported by ONR 321OA.]

9:10

3aAO4. Properties of the ambient noise field recorded at the 150 m isobath during the 2016–2017 Canadian Basin Acoustic Propagation Experiment. Jason D. Sagers and 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)

The Applied Research Laboratories at the University of Texas at Austin (ARL:UT) deployed two passive acoustic recording systems along the 150 m isobath of the Chukchi Shelf during the 2016–2017 Canadian Basin Acoustic Propagation Experiment (CANAPE). The first system was a single-hydrophone recorder located on the seafloor, while the Persistent Acoustic Observation System (PECOS) contained a horizontal line array of hydrophones along the seabed and a vertical line array spanning a portion of the water column. The systems were deployed and recovered during open-water conditions, but remained in place during the ice-formation, ice-covered, and ice-melt time periods. This talk presents initial findings of the statistical ambient noise levels during the year-long experiment, presents beam-noise levels recorded by PECOS, and qualitatively discusses the natural, biologic, and anthropogenic sounds present in the acoustic recordings. [Work sponsored by ONR.]

9:25–9:40 Break

Invited Paper

9:40

3aAO5. Acoustical characteristics and contributions of bubbles released from melting glacier ice. Matthew C. Zeh, Preston S. Wilson, Kevin M. Lee (Dept. Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, 3374 Lake Austin Blvd., Apt. B, Austin, TX 78703, mzeh@utexas.edu), and Erin C. Pettit (Dept. of GeoSci., Univ. of Alaska Fairbanks, Fairbanks, AK)

Glacierized fjords present a unique acoustic environment that is significantly louder than other ice-covered environments, with average sound pressure levels of 120 dB re 1 μPa with a broad peak between 1 and 3 kHz [Pettit *et al*., Geophys. Res. Lett. **42**, 2309–2316 (2015)]. The energy within this peak is due to the release of bubbles escaping from pressurized air-filled pores within melting glacier ice. These bubbles form during glacier formation via the compression of successive seasonal snow layers. During ice melt, the pressurized air cavities are released, jetting and squirting from the ice and into the surrounding ocean environment [Lee *et al*., Proc. Meetings Acoust. **20**, 070004 (2014)]. In order to more sufficiently characterize this dynamic physical process, acoustic measurements and high-speed video recordings were made using melting glacier ice samples taken from LeConte Glacier in southeastern Alaska. Samples were melted and recorded in a controlled laboratory apparatus as well as at a range of depths at Lake Travis northwest of Austin, TX. Insights gained from these tests are then compared to field recordings taken in LeConte Bay from October 2016 to September 2017. [Work supported by NDSEG Fellowship and ONR.]

Contributed Papers

10:00

3aAO6. Trends in wind, rain, and ships measured from ambient sound in the Bering Sea. Jennifer L. Miksis-Olds (School of Marine Sci. & Ocean Eng., Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, j.miksisolds@unh.edu) and Jeffrey Nystuen (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Advances in low-cost passive acoustic technology are providing long term, ambient sound time series from which to better understand the complex interactions between marine life, the environment, and mankind in remote, seasonally ice covered areas such as the Bering Sea. Remotely deployed Passive Acoustic Listeners (PALs) operate according to an adaptive sampling protocol as opposed to recording continuously, thus enabling data collection for a full year at a sampling rate of 100 kHz. Adaptive subsampling overcomes hardware and power limitations by reducing the amount of data acquired, while maintaining a high probability of detection and classification of targeted signals. Source contributions to the overall soundscape vary in space and time, and are highly dependent on temporal, geographic, and oceanographic factors. The differences observed in the acoustic record over time and between locations reflect changes in both physical and biological dynamics. This study focused only on the changing patterns of seasonal sea ice dynamics, storm frequency and intensity, and human activity in the acoustic record. As polar conditions continue to change due to reductions in sea ice, warming ocean temperatures, and increases in human use, passive acoustic monitoring will be a valuable tool to identify and track weather dynamics.

10:15

3aAO7. Studying melting icebergs with ambient noise oceanography. Oskar Glowacki (Inst. of Geophys. Polish Acad. of Sci., Ksiecia Janusza 64/413, Warsaw 01-452, Poland, oglowacki@igf.edu.pl), Grant B. Deane (Marine Physical Lab, Scripps Inst. of Oceanogr., La Jolla, CA), and Mateusz Moskalik (Inst. of Geophys. Polish Acad. of Sci., Warsaw, Poland)

Marine-terminating glaciers are retreating at an unprecedented pace, largely as a result of enhanced submarine melting. However, studying ice-ocean interactions is complicated due to both harsh conditions prevailing in glacial bays and lack of scientific methods enabling continuous measurements. Recent studies have shown that high underwater noise levels measured in the Arctic are related to glacier melt, but quantitative research requires proper separation of individual noise sources, including icebergs and glacier fronts. Therefore, we show results of field experiments carried out in 2013, 2015, and 2016 in Hornsund fjord, Svalbard, to present directionality and statistics of the noise produced by melting icebergs. Measurements of noise directionality were conducted with 3-hydrophone acoustic array. Calculated angles of arrivals for the noise at the frequency range of 1–10 kHz correspond well to locations of individual, grounded icebergs. The amplitude of sound emitted by these sources has a symmetric α-stable distribution, with parameters depending on separation between iceberg and receiver. These findings demonstrate that ambient noise oceanography is an efficient tool to detect and track icebergs using natural noise they produce during melting. [Work funded by the Polish National Science Centre, Grant No. 2013/11/N/ST10/01729 and by ONR, Grant No. N00014-17-1-2633.]

10:30

3aAO8. The vertical directionality of melt noise from a glacier terminus. Grant B. Deane (Marine Physical Lab., Univ. of California, San Diego, 13003 Slack St, La Jolla, CA 92093-0238, gdeane@ucsd.edu) and Oskar Glowacki (Polar Sci., Inst. of Geophys., Warsaw, Poland)

Measurements of the vertical directionality of the sound of bubbles released explosively from a melting glacier terminus are presented. These data are motivated by a desire to infer ice melt rates using the sound generated by bubbles, trapped in the glacier ice at the base of the fern layer and pressurized over time, as they are released by the melting terminus. The free energy available to generate noise is a function of the difference between the bubble internal gas pressure and hydrostatic pressure, both of which can vary with depth beneath the sea surface. Previous studies of bubble gas pressure in glacier ice suggest that the noise generated by bubble release should decrease with increasing depth below the surface. Measurements of noise directionality made with a compact, 4-element hydrophone array approximately 300 m in front of Hansbreen Glacier in Hornsund Fjord, Southwestern Svalbard in the summer of 2017 will be presented and discussed. This study suggests that only the top few 10's of meters of ice cliff generate the majority of melt noise. This result has important implications for the interpretation of bubble noise in terms of ice melt rate. [Work funded by ONR, Grant No. N00014-17-1-2633 and by the Polish National Science Centre, Grant No. 2013/11/N/ST10/01729.]

WEDNESDAY MORNING, 9 MAY 2018 GREENWAY F/G, 9:00 A.M. TO 12:00 NOON

Session 3aBA

Biomedical Acoustics and Physical Acoustics: Induction Mechanisms for Bubble Nucleation I

Jeffrey B. Fowlkes, Cochair
Radiology, Univ. of Michigan, 3226C Medical Sciences Building I, 1301 Catherine Street, Ann Arbor, MI 48109-5667

Ronald Roy, Cochair
Engineering Science, University of Oxford, Parks Road, Oxford OX1 3PJ, United Kingdom

Chair's Introduction—9:00

Invited Papers

9:05

3aBA1. Homogeneous vs heterogeneous nucleation, a comparison. Charles C. Church (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38677, cchurch@olemiss.edu)

Vapor bubbles of sufficient size to serve as effective acoustic scatterers or cavitation nuclei may form in materials having regions of very low interfacial tension when exposed to an acoustic wave. These regions may be simple "hot spots," *i.e.*, collections of molecules having higher-than-average kinetic energies which therefore have lower surface tension than is found a room temperature, or the interfaces between dissimilar materials, *e.g.*, the lipid coating on a microdroplet of liquid perfluourocarbon used as an ultrasound contrast agent. In addition, the curvature of such interfaces can act to focus an impinging acoustic beam in such a way as to increase its rarefactional pressure above that of the incoming wave. Another factor not often considered is that any foreign material, whether dissolved liquid or minute solid particulate, can act to disrupt the normal molecular structure of the liquid, thereby providing local regions of reduced interfacial tension. The energetics of the nucleation process shows that the rate of nucleation events is highly sensitive to the instantaneous temperature and moderately sensitive to both the magnitude and the duration of the rarefactional pressure of the acoustic wave. These and additional factors influencing nucleation processes will be discussed.

9:25

3aBA2. Histotripsy—From nucleation of bubbles to tissue disruption. Jeffrey B. Fowlkes (Radiology, Univ. of Michigan, 3226C Medical Sci. Bldg. I, 1301 Catherine St., Ann Arbor, MI 48109-5667, fowlkes@umich.edu)

Histotripsy utilizes focused ultrasound to nucleate gas bubbles deep within tissue and the associated bubble activity to induce biological effects at the level of tissue disruption/emulsification. Ultrasonic fields for inducing such effects include both pulses with as little as a single predominant negative half cycle to pulses with thousands of cycles and more. Different ensembles of bubbles leads to differing proposed mechanisms for tissue damage. Control of the damaged area is due to the threshold phenomena associated with the nucleation and the large focal gain in the main lobe, where the tissue disruption can be confined to fractions of the lateral beam width with little collateral damage. In addition, manipulation of the residual nuclei from a given ultrasound pulse can control cavitation events in subsequent pulses to allow greater and more uniform effects. The degree of tissue damage is equally impressive where disruption can be down to the subcellular level. At the same time, ultrasound parameters can be adjusted to selectively spare larger vessels from damage or retain the extracellular matrix. This variety of histotripsy properties opens up numerous potential medical applications, which will be discussed along with the conditions needs to achieve controlled and effective tissue disruption.

9:45

3aBA3. Nucleation and stabilization of cavitation nuclei in liquids. Lawrence A. Crum (APL, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, lacuw@uw.edu) and Julianna C. Simon (Acoust., Penn State University, University Park, PA)

The presence of cavitation nuclei are normally required in order to induce cavitation in water-based fluids, as the homogeneous nucleation threshold is beyond the range of most acoustic-pressure generation systems (a notable exception is the "intrinsic threshold" achieved by some histotripsy devices). There are a number of potential models for these nuclei, but one that continues to receive favor is the "crevice model," in which a pocket of gas is contained in a crack or crevice in a contaminating particle and then stabilized against diffusion by the geometry of the crevice. This presentation will describe this model in some detail, present some (old) data that support the model, and some new data that imply that such activatable nuclei exist in mammalian tissue.

10:05

3aBA4. The induction of inertial cavitation from polymeric nanocups—From theory to observation. James J. Kwan (School of Chemical and Biomedical Eng., Nanyang Technolog. Univ., 62 Nanyang Dr., Block N1.2, 01-06, Singapore 637459, Singapore, james-kwan@ntu.edu.sg), Guillaume Lajoinie (Phys. of Fluids, Univ. of Twente, Enschede, Netherlands), Eleanor P. Stride (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), Michel Versluis (Phys. of Fluids, Univ. of Twente, Enschede, Netherlands), and Constantin Coussios (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom)

The inability for therapeutics to distribute throughout the entirety of the tumor is a major challenge in cancer therapy. Acoustic cavitation from microbubbles promotes drug distribution and improves therapeutic efficacy. Yet microbubbles are too large to navigate the microvasculature of the tumor, and are destroyed by the ultrasound wave. Thus, there is a need for submicron cavitation nucleation agents. Recently, we have developed a submicron polymeric nanocup capable of trapping a nanobubble within the surface crevice. Using a modified Rayleigh-Plesset model that accounts for the size and shape of the crevice on the surface of the nanocup, we predicted that the cavity trapped bubble expands and contracts yet remains in the cavity. With a sufficient peak negative pressure amplitude, the model indicated that the surface bubble extends beyond and detaches from the crevice before inertial collapse. To verify our predictions, we exposed nanocups to high intensity focused ultrasound at different driving frequencies and observed bubble nucleation from nanocups using the Brandaris ultra high-speed camera. Acoustic emissions were recorded using a passive cavitation detector. These direct observations of the induction of inertial cavitation from nanocups verified the predictions made by the modified Rayleigh-Plesset crevice model.

10:25–10:40 Break

10:40

3aBA5. The effect of hypobaric pressure on the kidney stone twinkling artifact. Julianna C. Simon (Graduate Program in Acoust., The Penn State Univ., Univ. Park, PA and Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Penn State, 201E Appl. Sci. Bldg., University Park, PA 16802, jcsimon@psu.edu), Oleg A. Sapozhnikov (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA and Dept. of Acoust., Moscow State Univ., Moscow, Russian Federation), Wayne Kreider, Michael Breshock (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), James C. Williams (Dept. of Anatomy and Cell Biology, Indiana Univ. School of Medicine, Indianapolis, IN), and Michael Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA)

Recently, our group proposed the color Doppler ultrasound twinkling artifact originates from stable crevice bubbles on the kidney stone surface because overpressure suppressed twinkling on *ex vivo* stones (Lu *et al., Ultrasound Med. Biol.* 2013). However, the hypothesis is not fully accepted because the bubbles were not directly observed. Here, the submicron-sized bubbles predicted by the crevice-bubble hypothesis are enlarged in *ex vivo* kidney stones by exposure to a pre-focal, off-axis lithotripter pulse ($p_+ = 1.5$ MPa, $p_- = 2.5$ MPa) or hypobaric static pressures (0.021 MPa, absolute) to simultaneously capture their appearance by high-speed photography and ultrasound imaging. In rough stones that twinkle, consecutive lithotripter pulses caused more than 50% of bubbles to grow reproducibly from specific locations on the stone surface, suggesting the bubbles were pre-existing. Conversely, on smooth stones that did not twinkle, repeated lithotripter pulses initiated bubbles from varying locations on the stone surface. Similarly, upon exposure to hypobaric static pressures, the simple expectation that twinkling would increase by enlarging bubbles largely held for rough-surfaced stones but was inadequate for smoother stones. These results suggest a correlation between kidney stone surface topography or stable surface crevice bubbles and twinkling. [Work supported by NSBRI through NASA NCC 9-58 and NIH DK043881.]

11:00

3aBA6. Laser nucleation of single bubbles and clouds in an acoustic resonator via pressure-dependent dielectric breakdown. R. G. Holt (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, rgholt@bu.edu), Jonathan R. Sukovich (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI), Phillip Anderson (Listen, Inc., Boston, MA), Ashwinkumar Sampathkumar (Dayton Res. Ctr., Riverside Res., New York, New York), Todd W. Murray (Mech. Eng., Univ. of Colorado, Boulder, CO), and D. F. Gaitan (Sound Eng., Univ. of San Buenaventura, Medellin, Colombia)

Obtaining bubbles on demand at precise times and locations in a non-contact fashion can be useful in a variety of applications. Of special importance is the combination of laser nucleation with acoustics, so that bubbles are only just nucleated by the optics but grown to macroscopic size solely by the acoustics. We present theory and experiment for the non-thermal laser nucleation of bubbles in an acoustic field in the absence of significant absorbing/scattering particles. First we present theory and experiment for the threshold for dielectric breakdown in water, resolving the distinct minimum at 20 bar. Then, we present a method and results for nucleating single and multiple bubbles with temporal uncertainty of 5 ns, and spatial uncertainty of 1 mm. Results for bubble number and first cycle expansion are reported as functions of the timing of the nucleating laser pulse with respect to the acoustic field. [Work supported by Impulse Devices, Inc.]

11:20

3aBA7. Micro-cavitation on demand via the nanoparticle mediated interaction of light and sound. Ronald Roy (Dept. of Eng. Sci., Univ. of Oxford, Parks Rd., Oxford OX1 3PJ, United Kingdom, ronald.roy@hmc.ox.ac.uk), Caleb H. Farny (Mech. Eng., Boston Univ., Boston, MA), Tianming Wu (Radiation Oncology, Univ. of Chicago, Chicago, IL), R. G. Holt (Mech. Eng., Boston Univ., Boston, MA), and Todd W. Murray (Mech. Eng., Univ. of Colorado, Boulder, CO)

The safe utilization of controlled cavitation for HIFU therapy and ultrasound assisted drug delivery requires nucleation sites for bubble formation. We consider the potential for nucleating transient vapor cavities using laser-illuminated gold nanoparticles combined with high-intensity focused ultrasound. An transparent polyacrylamide gel phantom was seeded with 82-nm diameter gold particles and exposed to 20 ns pulses from a 532 nm Nd:Yag laser. Laser firing was synchronized with the arrival of a burst of 1.1 MHz focused ultrasound. Acoustic emissions from ensuing inertial cavitation were detected passively using a 15 MHz focused transducer. At a laser energy of 0.10 mJ/pulse, the resulting inertial cavitation nucleation threshold pressure (peak-negative focal pressure) was as low as 0.92 MPa. In comparison, a peak-negative focal pressure of 4.50 MPa was required to nucleate detectable cavitation without laser illumination (nano-particles were present in both cases). Experimental results agree well with a simple model for transient heating and cavity formation. Since the particles are durable, one can re-activate them as needed, essentially yielding cavitation nuclei "on demand." [Work supported by the Dept. of the Army (Award No. DAMD17-02-2-0014) and the Center for Subsurface Sensing and Imaging Systems (NSF ERC Award No. EEC-9986821).]

11:40

3aBA8. Nucleation pressure threshold in acoustic droplet vaporization. Christopher Miles, Charlie Doering, and Oliver D. Kripfgans (Dept. of Radiology, Univ. of Michigan, Ann Arbor, MI 48109-5667, oliver.kripfgans@umich.edu)

Acoustic droplet vaporization (ADV) has been studied by several laboratories around the world, including the parametric dependence of its nucleation. In this abstract, we will present our approach of combining classical homogeneous nucleation theory with superharmonic focusing to predict necessary pressures to induce nucleation in ADV. We will show that linear acoustics is a valid approximation to leading order when particle displacements in the sound field are small relative to the radius of the exposed droplet. This is done by perturbation analysis of an axisymmetric compressible inviscid flow about a micrometer-sized droplet with small surface perturbations relative to the mean droplet radius when subjected to an incoming ultrasonic wave. A calibrated single element spherical focus transducer (Olympus A321S 7.5 MHz) was positioned in a water tank above an inverted microscope (Leica DM IL) to emit single ten-cycle bursts. Single droplets (3–30 μm) were placed on ultrasound gel and exposed quasi free-field. Theoretical derivations and predictions will be compared to experimental findings. The ADV nucleation pressure threshold inside the droplet is calculated to be -9.33 ± 0.30 MPa for typical experimental parameters. As a result we are able to predict if a given incident pressure waveform will induce nucleation in an exposed perfluorocarbon droplet.

Session 3aEAa

Engineering Acoustics: General Studies on Structures

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

Contributed Papers

8:15

3aEAa1. Application of acoustic fabrics to improve the insertion loss of a partial enclosure. Weiyun Liu and David Herrin (Dept. of Mech. Eng., Univ. of Kentucky, 151 Ralph G. Anderson Bldg., Lexington, KY 40506-0503, weiyun.liu@uky.edu)

The sound absorption coefficient, transmission loss, and transfer impedance of different sound absorbing fabrics are measured. It is shown that the properties are similar to microperforated panels. Using microperforated panel equations, effective hole diameter and porosity are determined via a least squares curve fit. The effective parameters can then be used to predict the sound absorptive performance for different cavity depths. It is demonstrated that the fabrics have good sound absorptive performance over a wide range of frequencies. The fabric was then positioned in a partial enclosure. Single and double layers of the fabric were placed over an opening and the insertion loss as a result of fabric application was measured. The insertion loss is increased by approximately 5 dB if two layers of the fabric is used.

8:30

3aEAa2. Determination of the inverse, blocked, and pseudo forces for an air compressor bolted to a steel plate. Keyu Chen and David Herrin (Dept. of Mech. Eng., Univ. of Kentucky, 151 Ralph G. Anderson Bldg., Lexington, KY 40506-0503, keyu.chen@uky.edu)

For many machines, input forces cannot be directly measured, but representative forces may be determined using inverse approaches. Three approaches have been suggested in the literature for the determination of representative forces. The different approaches are described and then applied to an air compressor bolted to a steel plate. The three approaches are utilized to determine the inverse, blocked, and pseudo forces, respectively. The calculated indirect forces are then used to determine the response on the steel plate, and there is a good agreement between all three methods. The calculated forces are then used to predict the effect of a boundary condition modification to the steel plate.

8:45

3aEAa3. Analytical and experimental methods for the non-destructive acoustic testing of fluid-filled critical infrastructure from the 19th century to the present. Harrison Richarz (ICONAC / VESI Boesman, 76 Roxborough Ln., Thornhill, ON L4J 4T4, Canada, harrison@iconac.co)

From boreholes and chimneys, to pipelines, boilers, storage tanks, reactors, and water mains, non-destructive acoustic tests of fluid-filled conduits and shells help to monitor vital infrastructure across the globe. The theoretical treatment of wave propagation in such structures has its origin in the 19th century. In the intervening years, advances in sensors, data logging, and signal processing led to practical applications, first in oil exploration and later in other fields such as medicine and water distribution. This commercial activity accelerated the understanding of wave propagation in large scale multi-layered and multimaterial structures to create a host of new commercial applications. The paper explores the evolution and current state of these technologies and the acoustics behind them through an historical study of the progression of conduit wall impedance modeling and the problems of reflection and dispersion, as encountered in borehole and pipe inspection applications.

9:00

3aEAa4. Spatiotemporal modulation for mid-air haptic feedback from an ultrasonic phased array. Brian Kappus (Ultrahaptics, 2479 East Bayshore Rd., Palo Alto, CA 94303, brian.kappus@ultrahaptics.com) and Ben Long (Ultrahaptics, Bristol, United Kingdom)

<!–StartFragment–>A tactile sensation can be experienced by focusing airborne ultrasound using a phased array. Nonlinear acoustic pressure alone is difficult to perceive so traditional methods use amplitude modulation in the range of 10–300 Hz to stimulate nerves on the hand most sensitive to those frequencies. We demonstrate that through rapid translation of focus points similar results can be obtained using spatiotemporal modulation. This allows for volumetric sensations to be created using maximum power possible from the array. This is made possible through a solving approach using a pseudo-inverse of the activation matrix followed by a power iteration eigenvalue solution. Advantages versus amplitude modulation and consequences for parametric audio will be discussed.<!–EndFragment–>

9:15

3aEAa5. Boundary condition effects of porous material in absorption measurements: Comparison of two impedance tubes. Bárbara Fengler, William D. Fonseca, Paulo Mareze, Eric Brandao (Acoust., UFSM, Santa Maria, RS, Brazil), and ARTUR Zorzo (Acoust., UFSM, SQN 315 BLG Apto 205, Asa Norte, Brasilia 70774070, Brazil, arturzorzo@hotmail.com)

Porous materials are widely used for acoustical treatment and insulation of various environments. Its main feature is acoustic absorption, yielding the acoustical absorption coefficient over frequency. One of the ways to extract this parameter experimentally is with impedance tube (or Kundt's Tube) measurement via the transfer function method. However, the compression of the sample or the existence of gaps between the sample and the tube causes of uncertainty in the experiment. This research focuses upon the study of these effects. In order to analyze the influence of the boundary condition in the measurements, they were performed into two impedance tubes of slightly different internal diameters (both have the maximum frequency of analysis up to 6.4 kHz, approximately). Two different porous materials with samples of the same thickness were characterized using both tubes. In this situation, the procedure forces the influence of lateral leakage or compression of the samples against the tube to arise. Thus, this allows the analysis on the effect of the material structure in the acoustical absorption coefficient.

9:30

3aEAa6. Structural vibration damping of a vacuum-assisted toilet. Michael T. Rose, Joshua F. Kilts (Phys., Brigham Young Univ., Brigham Young University, Provo, UT 84602, rose.michael@byu.edu), B. Dagan Pielstick (Mech. Eng., Brigham Young Univ., Provo, UT), Scott D. Sommerfeldt, Kent L. Gee (Phys., Brigham Young Univ., Provo, UT), and Scott L. Thomson (Mech. Eng., Brigham Young Univ., Provo, UT)

Vacuum-assisted toilet noise can be unsettling and even uncomfortable. One common way to reduce noise levels is to damp structural vibrations that radiate sound. We investigated whether constrained layer damping (CLD) treatments could reduce the radiated noise level on a vacuum-assisted toilet. To find the structural anti-node locations of a toilet, we excited an airplane toilet with a shaker and scanned the velocity response of the inside of the bowl with a 3-dimensional scanning laser Doppler vibrometer (3D SLDV). We also scanned the bowl with an accelerometer during a repeated flush cycle. A microphone placed one meter above the bowl measured the radiated sound level. We applied 3M 4014, 3M 2552, Pyrotek Decidamp CLD, and Velcro individually to the bowl and determined the reduction in structural vibrations and sound radiation. The bowl's rim on the front vibrated the most. Structural vibrational energy concentrated around 100-400 Hz while radiated sound concentrated around 400 Hz–2 kHz. Applying damping materials reduced structural vibrations, sometimes by 20 dB. We conclude that CLD treatments are able to reduce structural vibrations. Further results of the investigation will be shown and discussed.

WEDNESDAY MORNING, 9 MAY 2018 GREENWAY D, 10:15 A.M. TO 11:45 A.M.

Session 3aEAb

Engineering Acoustics: General Studies on Transducers

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

3a WED. AM

Contributed Papers

10:15

3aEAb1. Real-time source localization using phase and amplitude gradient estimator for acoustic intensity. Jacey G. Young (Dept. of Phys., St. Norbert College, 100 Grant St., Ste. 2324, De Pere, WI 54115, jacey.young@snc.edu), Joseph S. Lawrence, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Recent demonstrations of real-time source localization systems have used a variety of direction-finding methods. One such system uses a multi-microphone probe to estimate acoustic intensity [Inoue *et al*., Proc. Meetings Acoust. **29**, 025001 (2016)], although the bandwidth of intensity estimates are traditionally limited by the microphone spacing. The Phase and Amplitude Gradient Estimator (PAGE) method [Thomas *et al*., J. Acoust. Soc. Am. **137**, 3366–3376 (2015)] greatly extends the bandwidth of active intensity, providing an accurate method of source localization for a wide bandwidth. An initial system implemented in LabVIEW estimates active intensity using the PAGE method to identify the direction of the source location in real time. Using a paired webcam, the program then overlays the direction as a three-dimensional arrow onto a webcam image. A demonstration will be provided. Initial quantitative results show the accuracy of the system and limitations of the device. [Sponsored by the NSF REU program.]

10:30

3aEAb2. Focusing airborne acoustic vortex beams by means of active diffraction gratings. Ruben D. Muelas-Hurtado, Joao L. Ealo (School of Mech. Eng., Universidad del Valle, Calle 13 No100-00, Cali, Valle del Cauca 76001, Colombia, ruben.muelas@correounivalle.edu.co), Jhon F. Pazos-Ospina (Escuela Militar de Aviación - EMAVI, Colombian Air Force, Cali, Colombia), and Karen Volke-Sepulveda (Instituto de Física, Universidad Nacional Autónoma de México, Mexico City, Mexico D.F., Mexico)

New technologies to generate acoustic vortex beams (VB) have received increasing attention due to their interesting capabilities to transfer angular momentum to matter and to attract particles to the propagation axis. One of the recently reported methods to generate acoustic VB is the fabrication of active diffraction gratings, i.e., acoustic sources whose geometry and vibration mode emulate the field obtained after passing a plane wave through a given passive grating [1]. In this work, we extend this approach by using an active spiral zone plate to produce a beam with a helical and focused structure. This is obtained by gluing an electroactive ferroelectret film on top of a lower electrode structured on a printed circuit board, which in turn results easy and reliable. The broadband feature of the film makes it possible to generate VB throughout the range of interest for most applications in air. Experimental results are compared with numerical simulations. The proposed ultrasonic source opens up new possibilities for particle manipulation and rotation control of small objects. [1] R. D. Muelas H, J. F. Pazos-Ospina, and J. L. Ealo, "Active diffracting gratings for vortex beam generation in air," J. Acoust. Soc. Am. **141**(5), 3569–3569, 2017.

10:45

3aEAb3. Calibration and acoustic performance of sparse transducer array. Lane P. Miller, Stephen C. Thompson (Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, lpm17@psu.edu), and Andrew Dittberner (GN Hearing, Glenview, IL)

In a previous research, a theoretical approach has been taken to predict, form, and optimize acoustic beam patterns from a sparse array of transducers. The current objective is to compare these theoretical results to experimentally obtained results from an assembled sparse array of transmit transducers. The array design and calibration procedure is presented. Acoustic beam patterns of the assembly are measured and compared to the theoretical model. Limitations of the array's performance are evaluated.

11:00

3aEAb4. Study on multi-channel speakers for controlling sound field of flat panel display. SungTae Lee (LG Display, Paju, South Korea), Hyung Woo Park, Myungjin Bae (IT, SoongSil Univ., 1212 Hyungham Eng. Bldg. 369 Snagdo-Ro, Dongjak-Gu, Seoul, Seoul 06978, South Korea, pphw@ssu.ac.kr), and Kwanho Park (LG Display, Paju, Korea (the Republic of))

Technological improvement in display industry, the organic light-emitting diode(OLED) panel manufacturer changed the quality of picture of the

screen from simple high definition to an augmented-reality or a virtual-reality. And OLED that construct with the actual pixels and emit light by themselves are becoming widespread, making the thickness thinner and the bezel thinner or even disappearing. However, these technology improvement made it difficult to achieve sufficient performance in a small space for sound quality, which is less important component of TV. In this study, we have developed a multi-channel flat panel speaker which is able to synthesis the sound field. That flat panel speaker consist with direct drive actuator and diaphragm as an outer—glass layer of OLED. This direct drive actuator speakers are used to realize multi-channel sound from a single OLED panel. Multi-channel speakers can control the sound field to set the direction of sound. This feature is a good condition for implementing mixed-reality. In previous research, it was confirmed that the viewer 's sense of reality increased when the position of sight and hearing were matched. Furthermore, we have been developed to increase the mixed reality, to improve the acoustic performance by adding directionality to the sound generated by the panel.

11:15

3aEAb5. A study on the sound generation at digital signage using OLED panel. Hyung Woo Park, Myungjin Bae (IT, SoongSil Univ., 1212 Hyungham Eng. Bulding 369 Snagdo-Ro, Dongjak-Gu, Seoul, Seoul 06978, South Korea, pphw@ssu.ac.kr), SungTae Lee, and Kwanho Park (LG Display, Paju, Korea (the Republic of))

With the development of ICT technology and convergence of media and ICT, smart media is spreading to people. The advertising industry through this smart screen is a rapidly growing future industry in the world. A new paradigm of bidirectional/customized features advertisement has become provided through IP-based new media such as smart TVs, portable smart devices, the Internet, IPTV (VOD), and digital signage. As one of the smart advertising industries, the digital signage is attracting attention as the fourth screen after TV, PC, and mobile with indoor/outdoor advertisement using information display. Advertising using the digital signage is installed in a lot of floating population place such as a subway station, a bus stop, an elevator, etc. Digital signage is developed with the display technology and the convergence of IT technology, but the development of appearance and screen configuration is mainly made. There are characteristics of high efficiency when communicate information with visual and auditory. However, up until now, digital signage has been difficult to provide vision and sound to people at the same time. The sound must be played through the screen; nevertheless, the smart screen does not pass the sound well, because of hardware. In this study, we introduce the technology that can generate direct sound using the characteristics of organic light-emitting diode (OLED) panel and introduce digital signage that can transmit video and sound simultaneously by using direct sound generation.

11:30

3aEAb6. A new lumped parameter model for the design of the free-flooded ring transducer. Kyounghun Been, Seungwon Nam (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), PIRO 416, Hyoja-dong, Nam-gu, Pohang-si, Gyeongbuk 790-784, South Korea, khbeen@postech.ac.kr), Haksue Lee, Hee-seon Seo (Agency for Defense Development, Changwon, South Korea), and Wonkyu Moon (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), Pohang, Kyungbuk, South Korea)

The free-flooded ring (FFR) transducer is the well-known low-frequency sound sources in underwater because of its broad operating frequency bandwidth and relatively small size. Many previous studies have been performed that predict the characteristics of an FFR transducer using an equivalent circuit model (ECM), a type of lumped parameter model (LPM) because an ECM is widely used to understand the properties of such transducers in the design process. However, it is quite difficult to predict the characteristics of an FFR transducer because the acoustic field is generated from its top and bottom openings, connected by the inner fluid, as well as the cylindrical ring surface. In this study, the authors investigated an ECM of an FFR transducer. The ECM consists of three parts: the piezoelectric ring, the cylindrical cavity, and the radiation load. In addition, an LPM which can consider mutual radiation loads was proposed to improve the accuracy of the model. The proposed models were compared and verified using commercial finite element method (COMSOL Multiphysics). We confirmed that LPM can predict characteristics of FFR transducer more accurately than ECM. [This work was supported by the National Research Foundation of Korea (NRF) grant funded by the Korea government (MSIP) (Grant No. 2016R1E1A2A02945515).]

Session 3aEDa

Education in Acoustics: Developing and/or Using Interactive Simulations for Teaching Acoustics

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

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

Invited Papers

8:00

3aEDa1. Use of interactive simulations in pre-class learning activities. Tracianne B. Neilsen (Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu)

To facilitate an effective in-class discussion, students must engage with the material prior to class. Assigned textbook reading is the traditional way students are asked to prepare, with possibly a reading quiz due before class. While carefully designed reading quizzes can be effective, a different approach is to use unscripted pre-class learning activities that provide hands-on interaction with the material. I created pre-class learning activities for the "Descriptive Acoustics" class at Brigham young University that use interactive simulations available on the internet. Provided with basic instructions, the students explore the simulation and write a description of their experience using terminology from the corresponding textbook chapter. Exploration followed by writing, even if incorrect, has increased the level of student understanding and provides a natural way to have students participate in class as they describe their experiences. The pre-class exposure to the interactive simulations also increases the efficacy of using the simulations during class, as each student is familiar with the interface and thus can better follow the in-class demonstration. Interactive simulations are a powerful active-learning tool for the current generation of students but only if they have had personal experience with the simulation prior to seeing it in class.

8:20

3aEDa2. Using interactive simulations to build understanding and promote scientific inquiry. Andrew A. Piacsek (Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926, andy.piacsek@cwu.edu)

Computer simulations of physical phenomena, with graphical output, have long been used to enable visualizations of processes that are inherently invisible, such as sound wave propagation, or that operate on time scales that make direct observation difficult. For students struggling to make sense of an abstract physical concept, such as the relationship between the propagation of a mechanical wave and the associated motion of the medium, an animated simulation of wave motion can accelerate the "aha!" moment of understanding. But to solidify and build on that understanding, students need to be able to formulate and test predictions. This can be accomplished with an *interactive* simulation, in which the user can adjust certain parameters of the model and view the response in the solution immediately. In this way, students can conduct virtual experiments. This presentation will use three examples of interactive simulations to demonstrate how interactive simulations can be used in a structured way to support specific learning objectives related to acoustics, and how they can be used to foster independent learning and scientific inquiry. Two examples come from a general education course in musical acoustics; the third example has been used in an advanced undergraduate/first-year graduate course in acoustics.

8:40

3aEDa3. Interactive (adjustable) plots and animations as teaching and learning tools. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, drussell@engr.psu.edu)

Interactive plots, such as are made possible with the Manipulate[ ] command in *Mathematica,* can be useful as teaching tools in the classroom, as well as learning aids for students outside of class. The use of adjustable sliders to change parameter values for an equation or system of equations allows for quick visual exploration of the effects of those changes on the resulting plot. Sometimes this visualization can help students understand difficult conceptual meaning hidden in mathematical expressions. Similarly, an adjustable interactive animation can effectively facilitate an understanding of concepts that are sometimes difficult to grasp from words or equations, or even "fixed" animations. This talk will demonstrate the creation of interactive plots and animations, and showcase several examples which the author has developed and used for teaching acoustics at the graduate and undergraduate levels. In addition, we will also discuss ways that students can be encouraged to create their own interactive plots or animations to aid in their own understanding.

9:00

3aEDa4. Simulations in acoustics and vibration education: What is the appropriate ratio of "learning to make" and "learning to use" for undergraduate students? Preston S. Wilson (Mech. Eng., Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292, pswilson@mail.utexas.edu)

Within most undergraduate engineering curricula in the US, there is room for perhaps one or two acoustics and vibrations-related courses, which are generally offered as technical electives rather than as required classes. Numerical computation is also taught in one or two (typically required) courses. Yet, in the current environment, very powerful commercial simulation packages are available that can be used to solve complex real-world problems. Since there is no room in the current curriculum to teach both aspects fully, this begs the question, "What is the most appropriate mix of teaching?" Should one focus on the fundamentals, such that students learn the basics of various solution methods, but suffer limited exposure to the solution of real-world applications, or should one focus on the use of specific software packages that can solve real-world problems? These issues are explored and various examples are presented in hopes to further discussion of this topic in the community.

9:20

3aEDa5. Using animations to better understand acoustic impedance. Brian E. Anderson (Dept. of Phys. & Astron., Brigham Young Univ., MS D446, Provo, UT 84602, bea@byu.edu)

It is very common for students to struggle to grasp a conceptual understanding of acoustic radiation impedance and input impedance. Often graduate students need exposure to impedance concepts in more than one course before they start to understand its importance and meaning. Instructors commonly suggest that impedance can be thought of as a resistance, ignoring the imaginary part of the impedance (the reactance), and the word "sloshing" is sometimes used to describe what the radiation reactance represents. This presentation will show some animations that have been developed in MATLAB to help students visualize the motion of the air particles near a vibrating surface. These animations utilize the relevant physical equations to observe the ratio of the pressure to the particle velocity. While these animations are not directly interactive, the user can change things like the frequency, source size, and excitation amplitude and recreate a new animation. Thinking of the impedance as the ratio of a potential quantity to a flow quantity can often be helpful, with this ratio having an in phase component and a component where the two quantities are 90 degree phase shifted.

9:40

3aEDa6. VSLM—The virtual sound level meter. Ralph T. Muehleisen (Energy Systems, Argonne National Lab., 9700 S. Cass Ave., Bldg. 221, Lemont, IL 60439, rmuehleisen@anl.gov)

Inexpensive sound level meters and low cost or free apps for smart phones and tables have made it much easier to introduce the concept of sound levels to a wide audience at very low cost. However, there are times when it is preferable to be able to take more complicated measurements that are generally only available on more expensive sound level meter. With fairly low cost digital recorders, one can now take a calibrated recording of an acoustic event and analyze the recording later. In this presentation, a free software package called VSLM: The Virtual Sound Level Meter is described. VSLM allows users to calibrate a recording and do many types of analysis including mimicking a sound level meter on fast, slow, Leq, octave, and 1/3 octave band analysis using fast FFT or digital meters that meet ANSI standards, high resolution spectral analysis, and create spectrograms.

10:00 – 10:20

10:20

3aEDa7. Numerical simulation and use of the PICUP framework for use in an undergraduate acoustics course. Andrew C. Morrison (Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431, amorrison@jjc.edu)

The use of numerical simulations in an undergraduate course introductory acoustics course can be a way to provide students with a method of exploring concepts that are difficult or costly to replicate in a classroom. For a course taught that serves a diverse group of students studying a large variety of majors, the importance of having multiple ways to present and revisit concepts should not be underestimated. Several simulations have been developed using Glowscript for use in a general education acoustics course. Additionally, the PICUP framework for development of exercise sets has been used to structure classroom activities in a way that encourages scaffolding of concepts from the most basic ideas to more complex applications. Examples of the Glowscript simulations and the development of PICUP exercises will be demonstrated.

10:40

3aEDa8. An interactive simulation of the filtered-x least-mean-squares algorithm for active noise control. Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu)

As part of a graduate acoustics course at Brigham Young University, students learn about principles of active noise control (ANC). One of the concepts covered is how a basic ANC algorithm functions. This presentation describes an interactive simulation of the filtered-x least-mean-squares (FXLMS) algorithm implemented in MATLAB. Available user-selected inputs include sampling frequency, type of signal to be controlled, number of filter coefficients, convergence coefficient, and system model parameters. As outputs, the students are able to observe the different signals, the attenuation, and the filter coefficients. Altering inputs and observing the simulation results motivates classroom discussion and facilitates students' gaining a working understanding of an ANC algorithm.

Contributed Paper

11:00

3aEDa9. Combining interactive simulation with scaffolded visualization in ESAIL to teach fundamental concepts of acoustical oceanography.
Jason E. Summers (ARiA, 1222 4th St. SW, Washington, DC 20024-2302, jason.e.summers@ariacoustics.com), Mark A. Ross, Zachary Walker, Paul Longerbeam, and Daniel Redmond (ARiA, Culpeper, VA)

To understand and interpret active-sonar returns and effectively employ active sonar systems for purposes such as antisubmarine warfare (ASW), operators must understand fundamental concepts of acoustical oceanography that govern how sonar interacts with the environment. Experts achieve high levels of performance in analysis and employment not by solving acoustic-propagation equations, but rather by employing simplified mental models that characterize the functional relationships governing how the environment affects acoustic propagation. Thus, a key goal in educating operators is helping them to develop and use such mental models. Here, we describe the development and employment of ESAIL, the Environment for Surface ASW Interactive Learning, a learning platform that combines interactive simulation with scaffolded visualization to facilitate instructors and students in conducting and assessing the outcome of "what if…?" experiments. In ESAIL, users are scaffolded in performing "what if…?" experimentation by tools that automatically classify user-modified acoustical environments as well as predictive acoustical models. This capability for direct exploration of cause-and-effect relationships with scaffolding to help develop expectations facilitates operators actively learning the mental models and mental-simulation capabilities experts use to interpret displays and reason about the underlying tactical scenario to achieve more optimal employment. [Work supported by the Naval Sea Systems Command.]

WEDNESDAY MORNING, 9 MAY 2018 — MIRAGE, 9:30 A.M. TO 12:30 A.M.

Session 3aEDb

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

Peggy B. Nelson, Cochair
Univ. of Minnesota, Minneapolis, MN 55455

Keeta Jones, Cochair
Acoustical Society of America, 1305 Walt Whitman Rd., Suite 300, Melville, NY 11787

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 Minneapolis 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: Acoustics of Choirs and Vocal Ensembles

James P. Cottingham, Chair
Physics, Coe College, 1220 First Avenue NE, Cedar Rapids, IA 52402

Chair's Introduction—9:00

Invited Papers

9:05

3aMU1. Traditional polyphony: Multipart singing in world cultures. Paul A. Wheeler (ECE, Utah State Univ., 1595 N. 1600 E., Logan, UT 84341-2915, pawheeler21@gmail.com)

Vocal polyphony has been part of the western musical culture since the 11th century. There are other cultures in the world, however, that use multipart singing in their traditional music that is not based upon the European classical style with parallel thirds and triad chords. This paper will discuss the use of vocal polyphony in several well-established traditions in the world, such as the Aka Pygmies from Central Africa, *Canto a tenore* from Sardinia, and iso-polyphony of Albania. It will contrast polyphonic types like parallel, drone, canonic, ostinato, and heterophonic polyphony. It will also illustrate how sound quality differs from the European classical style. An understanding of polyphonic traditions can assist us in understanding people whose cultures differ from our own.

9:25

3aMU2. The relation between choir size and choir dynamics. Ingo R. Titze and Lynn M. Maxfield (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)

If six distinct levels of choir dynamics (*pp p mp mf f ff*) are to be achieved over two octaves of fundamental frequency in a choir section, how distinct are these levels? The just noticeable difference for sound level in a free field environment is 1-2 dB, while a doubling of loudness requires a 10 phon increase (10 dB SL increase at 1000 Hz). Most singers cannot double their vocal loudness 5 times. Generally the dynamic levels are 3–6 dB apart, depending on the individual *voice range profiles* of the singers. Overall choir size has no effect on dynamic range, unless the size is varied dynamically by not all singers singing all the time. A few loud voices dominate *ff* if everyone sings. To achieve an effective choir *pp,* only a few voices who can sustain very soft notes should sing. The dynamic range can be significantly limited when choral blend for loudness is imposed on a non-homogeneous group of singers.

9:45

3aMU3. Choir spacing vs choir formation: Long-term average spectra comparisons. James F. Daugherty (School of Music, Univ. of Kansas, 1530 Naismith Dr., Ste 448, Lawrence, KS 66045, jdaugher@ku.edu)

Choral methods books have long advised that repositioning choir singers solely according to voice part sung (i.e., choir formation) alters choral sound quality. To date, acoustical studies of choir formation have found no significant mean differences in long-term average spectra (LTAS) attributable to this strategy. Other investigations have indicated that changing the spatial distance between choir singers (i.e., choir spacing) yields significant LTAS differences. As yet, however, no study has compared the LTAS of choir spacing and choir formation conditions using the same choir. This paper reports experiments comparing LTAS of performances acquired from two microphone positions (conductor position, audience position) by three choirs of varied voicing (TTBB, SSAA, and SATB) singing in contrasting inter-singer spacing conditions (close, lateral, circumambient, and uneven) and voice-part formations (block sectional, mixed). Although recording venues and sung literature varied, each of the three choirs exhibited a pattern of significant mean timbral differences according to singer spacing conditions, but not according to formation conditions. Results were discussed in terms of the logics informing the spacing and formation approaches to modifying choir sound and implications for choral practice.

10:05

3aMU4. Effects of a straw phonation protocol on acoustic and perceptual measures of choirs. Jeremy N. Manternach (College of Education/School of Music, The Univ. of Iowa, 311 Communications Ctr., Iowa City, IA 52242, jeremy-manternach@uiowa.edu)

Voice instructors, choral directors, and voice professionals have long utilized semi-occluded vocal tract (SOVT) exercises to evoke efficient voicing from their students or clients. These exercises, which involve narrowing and/or lengthening the vocal tract as a means of increasing pharyngeal pressure, can include nasal consonants, lip or tongue trills, raspberries, or vocalizing through a tube or straw. Researchers have noted that the increased pressure reduces the amount of breath pressure required to initiate voicing (i.e., phonation threshold pressure) while maintaining or increasing acoustic output (e.g., sound pressure level, formant intensity). The result may be increased "vocal economy" or "vocal efficiency" (i.e., increased vocal output with decreased effort). Until recently, however, this

research was limited to effects on individuals. This paper includes a series of studies in which choirs of varied voicing (SATB, SSAA, and TTBB) sang unaccompanied pieces prior to and after performing voicing exercises through a small stirring straw. Long-term average spectra have indicated that most choirs maintained or increased acoustic output in the conglomerate, choral sound after the protocols. Singers have also perceived improved choral sound and decreased singing effort. These results may interest choral directors who wish to improve the vocal economy of their ensembles.

10:25

3aMU5. The barbershop sound: The characteristic timbre of the male barbershop quartet. Colin Drown (Phys. Dept., Truman State Univ., Kirksville, MO 63501, cdd5168@truman.edu) and James P. Cottingham (Phys., Coe College, Cedar Rapids, IA)

A feature of the homophonic barbershop style is an emphasis on major and dominant seventh chords which are sung with little vibrato and with pitches adjusted to achieve tuned chords in just intonation. A principal resource for the investigation reported here was the library of online recordings of the 2016 and 2017 international competitions sponsored by the Barbershop Harmony Society. These recordings provide a wealth of examples of the barbershop sound. They include hundreds of quartet samples performed in a context that features a uniform expectation of what constitutes good sound, along with uniform recording conditions and ranking by judges who serve as expert listeners. A large number of spectra were obtained for a loud, sustained major chord on an open vowel. Tuning of these chords was investigated, and for each sample a spectral contour was constructed with amplitudes normalized to simplify comparisons. From the consistency of these spectral contours it is possible to obtain a picture of the timbral features that distinguish the barbershop sound. In addition, patterns of relatively small differences in spectral envelope among the samples are strongly correlated with the ranking of the quartets by the competition judges.

WEDNESDAY MORNING, 9 MAY 2018 GREENWAY J, 9:00 A.M. TO 11:30 A.M.

Session 3aPA

Physical Acoustics: General Topics in Physical Acoustics I

Aaron Gunderson, Chair
The University of Texas at Austin, Austin, TX 78712

Contributed Papers

9:00

3aPA1. Development of a dual temporal-spatial chirp method for the generation of broadband surface acoustic waves. Marc Duquennoy, Dame Fall, Mohammadi Ouaftouh, Farouk Benmeddour, Salah-eddine Hebaz, Nikolay Samgin (Univ. Valenciennes, CNRS, Univ. Lille, YNCREA, Centrale Lille, UMR 8520 - IEMN, DOAE, F-59313, IEMN-DOAE, Campus Mont Houy, Valenciennes 59300, France, Salah-Eddine.Hebaz@univ-valenciennes.fr), Bogdan PIWAKOWSKI (Univ. Valenciennes, CNRS, Univ. Lille, YNCREA, Centrale Lille, UMR 8520 - IEMN, DOAE, F-59313, Villeneuve-d'Ascq, France), and Frederic JENOT (Univ. Valenciennes, CNRS, Univ. Lille, YNCREA, Centrale Lille, UMR 8520 - IEMN, DOAE, F-59313, Valenciennes, France)

This study deals with the optimization of transducers for Rayleigh-type Surface Acoustic Waves (SAW) generation. These transducers are based on Interdigital Transducers (IDT) and are specifically developed to characterize properties of thin layers, coatings, and functional surfaces. In order to characterize these coatings and structures, it is necessary to work with wide bandwidths IDT operating in the high frequencies. Therefore, in this study a spatial chirp-based on an IDT are realized for wideband SAW generation. The use of impulse temporal excitation (Dirac-type negative pulse) leads to a wide band emitter excitation but with significantly limited SAW output amplitudes due to the piezoelectric crystal breakdown voltage. This limitation can be circumvented by applying a temporal chirp excitation corresponding in terms of frequency band and duration to the spatial chirp transducer configuration. This dual temporal-spatial chirp method was studied in the 20 to 125 MHz frequency range and allowed to obtain higher SAW displacement amplitudes with an excitation voltage lower than that of the impulse excitation.

9:15

3aPA2. Acoustic vortex beam created with a continuous phase ramp lens and resulting radiation torque response of a sphere. Auberry R. Fortuner, Timothy D. Daniel, and Philip L. Marston (Phys. and Astronomy Dept., WSU, Washington State Univ., Pullman, WA 99164-2814, auberry.fortuner@wsu.edu)

An acoustic vortex beam was created in water using a lens with a phase ramp placed in front of a focused transducer. The lens was made from a polystyrene disk and CAD machine milled to have a continuous ramp in height about the center axis, with the ramp step height corresponding to a 2π phase difference between waves propagating in water and polystyrene at 500 kHz. This causes the transmitted beam through the lens to exit as a vortex beam with an angular phase ramp. It has been predicted [L. Zhang and P. L. Marston, J. Acoust. Soc. Am. **136**, 2917–2921 (2014)] that the torque exerted by a vortex beam on an impenetrable sphere in a viscous fluid is proportional to the incident intensity due to dissipation of angular momentum in the viscous boundary layer near the sphere. This should result in steady rotation of the sphere with viscous drag proportional to angular velocity. Our upward-directed vortex beam was used to trap and spin a Styrofoam (closed foam) ball floating at the water surface. The spin rate was approximately proportional to the square of the source voltage as expected. [Work supported by ONR.]

9:30

3aPA3. Experimental measurement and analysis of acoustic wave propagation in water-filled steel pipeline using the iterative quadratic maximum likelihood algorithm. Zhao Li, Liwen Jing, Wenjie Wang, Yue Li, Amartansh Dubey (Electron. and Comput. Eng., Hong Kong Univ. of Sci. and Technol., Rm 2427, Academic Bldg., Clear Water Bay, Kowloon, Hong Kong, Hong Kong 999077, Hong Kong, eelizhao@ust.hk), Pedro Lee (Dept. of Civil and Natural Resources Eng., Univ. of Canterbury, Christchurch, New Zealand), and Ross Murch (Electron. and Comput. Eng., Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong)

Acoustic wave propagation (up to 50 kHz) within a water-filled steel pipeline is studied using laboratory experiments. The experiments were carried out in a 6 m length of cylindrical stainless steel pipeline using acoustic transducers to acquire signals from 100 locations uniformly spaced along the longitudinal axis of the pipe. By applying the iterative quadratic maximum likelihood algorithm (IQML) to the experimental results, parameters such as wavenumbers, attenuations, and mode amplitudes were accurately extracted for individual modes from the measurement data. We found that the IQML algorithm could extract these parameters more accurately in situations where the measurement data had low signal to noise ratio as compared to other algorithms such as Prony's method. A very good match was obtained between the experimental results and predictions from an analytical waveguide model for the wavenumber dispersion curves, attenuations, and acoustic power characteristics of the axisymmetric and non-axisymmetric modes. Additional physical explanations of the propagation phenomena in the pipeline waveguide were obtained using the experimental results and analytical model.

9:45

3aPA4. Qualitative analysis of mode transitions in bottle-shaped resonators with waterfall plots. Bonnie Andersen, David Eldred, and Josh Dimond (Phys., Utah Valley Univ., MS 179, 800 W University Pkwy, Orem, UT 84057, bonniem@uvu.edu)

A closed bottle-shaped resonator consists of the coupled neck and cavity. Such a system yields avoided crossings where the resonance of the neck matches that of the cavity when one bottle dimension is varied. Self-sustained oscillations within the bottle are generated thermoacoustically. Mode transitions were previously observed to occur at the same position within a few millimeters when a piston controlled with a translation stage was moved with a manual control to adjust the cavity length. The dominant mode was recorded using a power spectrum of the signal measured with a pressure sensor. In this study, the piston motion is automated and eight neck/cavity combinations were tested at three different piston speeds and at various input powers. The input powers were adjusted to just above thermoacoustic onset and not to exceed thermal limits of the materials used. Waterfall plots allow the visualization of the time evolution of the power spectrum where intensity is plotted both as a function of time and frequency. Qualitatively, the transitions occur at the same place within the cavity after a threshold input power is reached. Interestingly, overtones appear in most cases to be harmonics of the fundamental with either all or only odd harmonics present.

10:00

3aPA5. Determining the mass of gas in large collection volumes with acoustic and microwave resonances. Keith A. Gillis, Jodie G. Pope, Michael Moldover (Sensor Sci. Div., National Inst. of Standards and Technol., 100 Bureau Dr., Mailstop 8360, Gaithersburg, MD 20899-8360, keith.gillis@nist.gov), and James B. Mehl (Sensor Sci. Div., National Inst. of Standards and Technol., Orcas, Washington)

We characterized a 1.8 m^3, quasi-spherical resonator, a pressure vessel informally known as the "big blue ball" or "BBB." The BBB will be the collection volume of a NIST-designed gas-flow standard that will operate at pressures up to 7 MPa. Using microwave resonance frequencies, we determined the volume of the BBB filled with argon as a function of pressure (up to 7 MPa) and temperature (near ambient). The measured pressure- and temperature-dependences of the BBB's volume are consistent with the published properties of carbon steels. We filled the BBB with 220 kg of argon in weighed increments of approximately 20 kg. We also determined the mass m_{acoust} of argon in the BBB by measuring the pressure, the acoustic resonance frequencies f_{acoust} of 3 modes, and using the known thermophysical properties of argon. The values of m_{acoust} were within $\pm$0.03 % of the gravimetrically determined masses. In a typical application, the BBB will never be isothermal. Nevertheless, the acoustic resonance frequencies quickly and accurately determine the average temperature and mass of the gas in the BBB despite significant, long-lived thermal gradients that are created by pressure changes. A theoretical model for effect of temperature gradients on the acoustic resonance frequencies is presented.

10:15–10:30 Break

10:30

3aPA6. Acoustical and optical resonances of two-sphere systems. Cleon E. Dean (Phys., Georgia Southern Univ., PO Box 8031, Math/Phys. Bldg, Statesboro, GA 30461-8031, cdean@georgiasouthern.edu)

Two sphere systems show strong morphology dependent resonances even for two spheres of identical composition [G. W. Kattawar and C. E. Dean, Opt. Letts. 8, 48–50 (1983).]. How do analogous fluid spheres respond in the analogous situations? Do they show similar or different resonances? Preliminary investigations suggested that the response of fluid bispheres do not show as rich a panoply of resonances [C. E. Dean and J. P. Braselton, JASA, **141**, 3735 (2017).]. How much do these systems differ? Would alternate boundary conditions restore some lost resonances? This talk attempts to answer these and other questions through the use of theoretical computational acoustics models.

10:45

3aPA7. Inferring atmospheric surface-layer properties from wind noise in the nocturnal boundary layer. Carl R. Hart (U.S. Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, carl.r.hart@usace.army.mil) and Gregory W Lyons (U.S. Engineer Res. and Development Ctr., Champaign, IL)

Uncertainties in the state of the atmosphere, to a large extent, limit the prediction accuracy of outdoor sound propagation. In particular, event sound propagation requires accurate knowledge of wind speed and temperature profiles, spatially averaged over the path of propagation. In a stable quasi-steady nocturnal boundary layer, wind speed and temperature gradients follow a scaling that is asymptotically independent of altitude and depends on Monin-Obukhov similarity parameters. Since these parameters describe the near-surface profiles of wind speed and turbulent intensity, which in turn are known to govern wind noise, it is anticipated that a connection exists between Monin-Obukhov parameters and the statistics of wind noise. It is expected that these parameters may be inferred from wind noise sensed by screened microphones. Ambient noise data collected in the southwest U.S. are analyzed for the purpose of examining whether Monin-Obukov similarity parameters may be inferred from wind noise. We explore the consequences of establishing inferences with a priori distributions for the similarity parameters, and utilizing wind noise data from microphones at one or more altitudes.

11:00

3aPA8. Absorption and dispersion of low-frequency sound waves in the main cloud layers of Venus. Adam Trahan and Andi Petculescu (Univ. of Louisiana at Lafayette, University of Louisiana at Lafayette, 240 Hebrard Blvd, Rm 103, Lafayette, LA, ajt6261@louisiana.edu)

We present predictions for the acoustic wavenumbers at low frequencies in the condensational cloud layers of Venus, occurring between 50 and 70 km, approximately. While the general thermodynamics of Earth clouds is well understood, that of Venusian clouds is still a matter of debate. Venus' clouds are formed primarily of H_2O and H_2SO_4 vapors and aqueous sulfuric acid droplets, the fluxes of which are not fully constrained due to the few *in situ* observations. This study is based on and adapted from the terrestrial model of Baudoin *et al.* (J. Acoust. Soc. Am. **130**, 1142 (2011)). Inside the clouds, the Navier-Stokes equations of continuum fluid mechanics are used for the gaseous (dry + vapor) and liquid phases of H_2O and H_2SO_4, combined with the equations describing the evaporation/condensation processes; the gaseous phase is treated as an ideal gas. Outside the clouds, in the CO_2—dominated atmosphere, the van der Waals equation of state is used.

[The work was supported by a Grant from the Louisiana Space Consortium (LaSPACE).]

11:15

3aPA9. Effect of clustered bubbly liquids on linear-wave propagation. Yuzhe Fan, Haisen Li, Chao Xu, and Baowei Chen (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145, Nantong St., Nangang District, Harbin, Heilongjiang 150001, China, fanyuzhe@hrbeu.edu.cn)

The spatial distribution of bubble liquids strongly affect linear-wave propagation in bubbly liquids. The classical understanding suggests that using positional correlations of the bubbles, such as "hole correction" or Perkus-Yevick approximation, describes the spatial information of bubble clouds. However, bubbly liquids are observed experimentally with complex bubble ensembles that take the form of clusters, filaments, and clouds which cannot be intuitively described by these pair-correlation functions and need to be better clarified. To achieve this purpose, a three-dimensional random model, the Neyman-Scott point process, is proposed to describe bubbly liquids with clustering. Base on this method, we study the influence of such phenomenon on acoustic dispersion and attenuation relations. A formula for effective wavenumber in bubbly liquids is derived, based on self-consistent method. Comparing with the equation of Commander and Prosperetti [J. Acoust. Soc. Am. **85**, 732 (1989)], our results show that the clustering can suppress peaks in the attenuation and the phase velocity as functions of the frequency. Further, we provide a numerical simulated method. A clustered bubbly liquid is simulated with strict mathematical method and the statistical informations are obtained through unbiased statistical approach. Through the results, we quantificationally analyze the influence of estimated value on predictions.

WEDNESDAY MORNING, 9 MAY 2018 NICOLLET D2, 7:55 A.M. TO 10:55 A.M.

Session 3aPPa

Psychological and Physiological Acoustics: Physiology Meets Perception

Sarah Verhulst, Cochair
Dept. Information Technology, Ghent Univ., Technologiepark 15, Zwijnaarde 9052, Belgium

Antje Ihlefeld, Cochair
Biomedical Engineering, New Jersey Institute of Technology, 323 Martin Luther King Blvd, Fenster Hall, Room 645, Newark, NJ 07102

Invited Paper

7:55

3aPPa1. The eardrums move when the eyes move: A multisensory effect on the mechanics of hearing. Kurtis G. Gruters (Psych. and Neurosci., Duke Univ., Fayetteville, NC), David L. Murphy, Cole D. Jenson (Psych. and Neurosci., Duke Univ., Durham, NC), David W. Smith (Psych., Univ. of Florida, Gainesville, FL), Christopher Shera (Caruso Dept. of Otolaryngol., Univ. of Southern California, Los Angeles, CA), and Jennifer M. Groh (Psych. and Neurosci., Duke Univ., LSRC Rm B203, Durham, NC 27708, jmgroh@duke.edu)

Interactions between sensory pathways such as the visual and auditory systems are known to occur in the brain, but where they *first* occur is uncertain. Here, we show a novel multimodal interaction evident at the eardrum. Ear canal microphone measurements in humans ($n = 19$ ears in 16 subjects) and monkeys ($n = 5$ ears in 3 subjects) performing a saccadic eye movement task to visual targets indicated that the eardrum moves in conjunction with the eye movement. The eardrum motion was oscillatory and began as early as 10 ms before saccade onset in humans or with saccade onset in monkeys. These eardrum movements, which we dub Eye Movement Related Eardrum Oscillations (EMREOs), occurred in the absence of a sound stimulus. The EMREOs' amplitude and phase depended on the direction and horizontal amplitude of the saccade. They lasted throughout the saccade and well into subsequent periods of steady fixation. We discuss the possibility that the mechanisms underlying EMREOs create eye movement-related binaural cues that may aid the brain in evaluating the relationship between visual and auditory stimulus locations as the eyes move.

Contributed Paper

8:15

3aPPa2. A model of the neuronal processing of interaural time disparities. Jörg Encke (Bio-Inspired Information Processing, Tech. Univ. of Munich, Boltzmannstrasse 11, Garching 85748, Germany, joerg.encke@tum.de), Florian Völk (WindAcoust., WinDC, Germany), and Werner Hemmert (Bio-Inspired Information Processing, Tech. Univ. of Munich, Garching, Germany)

The auditory system of humans and other mammals is able to use interaural time and intensity differences as well as spectral cues to localize sound sources. One important cue for localizing low-frequency sound sources in the horizontal plane are interaural time differences (ITDs), which are first analyzed in the medial superior olive (MSO) in the brainstem. Results from electrophysiological and psychoacoustic studies suggest ITD encoding in the relative activities of neuronal populations in the two brain hemispheres. This contribution first explores the neuronal encoding of fine-structure ITDs using a physiologically motivated spiking neuronal-network model of the mammalian MSO. Results from this model confirm robust ITD encoding in the relative activity of the MSOs in both hemispheres. Based on the neuronal-network simulations, a simple probabilistic model of subsequent ITD processing is proposed. This simplified model connects the neuronal responses of the two hemispheres with different hearing sensation properties, such as lateral position, spatial extent, or number of spatially separable sensations. First predictions from the model are discussed with regard to the results of a series of psychoacoustic experiments on the lateralization of pure-tone impulses presented via headphones.

Invited Papers

8:30

3aPPa3. Evidence for the origin of the binaural interaction components of the auditory brainstem response. Sandra Tolnai and Georg Klump (Animal Physiol. & Behaviour Group, Dept. for Neurosci., Carl von Ossietzky Univ. Oldenburg, Carl-von-Ossietzky-Str. 9-11, Oldenburg 26129, Germany, sandra.tolnai@uni-oldenburg.de)

The binaural interaction component (BIC) is discussed as a potential tool to objectively measure listeners' binaural auditory processing abilities. It is obtained from auditory brainstem responses (ABRs) by subtracting the sum of the ABRs to monaural left and monaural right stimulation from the ABR recorded under binaural stimulation. The sources of the BIC, however, have not yet been agreed upon. Candidate source regions are the lateral and medial superior olives (LSO and MSO, respectively) in the superior olivary complex where excitatory and inhibitory inputs converge. Our study aims at identifying the source of the BIC. Simultaneously to ABRs, we recorded local-field potentials (LFPs) and single-unit (SU) responses from the LSO and MSO of ketamine/xylazine-anaesthetised Mongolian gerbils and derived LFP-related and SU-related BICs the same way as ABR-related BIC. We then compared the properties of LFP-related and SU-related BICs with the ABR-related BICs. LFP-related BICs recorded in the LSO did not mirror the characteristics of the ABR-related BIC while the SU-related BIC did. In the MSO, neither LFP-related nor SU-related BIC mirrored ABR-related BICs. This suggests that the output of LSO units but not MSO units contribute substantially to the generation of the ABR-related BIC.

8:50

3aPPa4. Neural bases of complex sound perception: Insights from an avian speech mimic. Kenneth S. Henry (Otolaryngol., Univ. of Rochester, 601 Elmwood Ave., Box 629, Rochester, NY 14642, kenneth_henry@urmc.rochester.edu) and Laurel H. Carney (Biomedical Eng., Univ. of Rochester, Rochester, NY)

Many complex sounds including speech contain periodic fluctuations in signal level known as amplitude modulation (AM). Auditory-nerve fibers encode AM primarily through response synchrony to signal modulation. In contrast, neurons of the midbrain encode AM through both response synchrony and substantial changes in average discharge rate. Specifically, many midbrain neurons with band-pass modulation tuning show enhanced discharge activity in response to a limited range of modulation frequencies. Here, we describe several recent studies on the relative importance of envelope synchrony vs. average rate coding in the midbrain for AM perception. Behavioral and neurophysiological experiments were conducted in the budgerigar, an avian species with the unusual capacity to mimic speech. Budgerigars were found to exhibit human-like behavioral sensitivity to a variety of complex sounds including AM stimuli and synthetic vowels with triangular and natural spectral envelopes. Most neurons in the budgerigar midbrain showed band-pass modulation tuning. Whereas either response synchrony or average discharge rate was sufficient to account for behavioral performance in some cases, only response synchrony could explain behavioral thresholds for (1) detection of low modulation frequencies and (2) formant-frequency discrimination in noise. These results highlight the significance of midbrain envelope synchrony for perception of complex sounds. [R00-DC013792 and R01-DC001641.]

9:10

3aPPa5. Is there a relationship between perceptual difficulties and auditory functioning at the periphery and the brainstem in children suspected with auditory processing disorder? Sriram Boothalingam (Commun. Sci. and Disord., Univ. of Wisconsin-Madison, 1975 Willow Dr., Goodnight Hall, Rm 482, Madison, WI 53706, boothalingam@wisc.edu), Sangamanatha Veeranna (National Ctr. for Audiol., Western Univ., London, ON, Canada), Chris Allan (Commun. Sci. and Disord., Western Univ., London, ON, Canada), David Purcell, and Prudence Allen (National Ctr. for Audiol., Western Univ., London, ON, Canada)

The test battery typically used for the diagnosis of auditory processing disorder (APD) is highly heterogeneous, with an emphasis on the central auditory nervous system. As such, the peripheral auditory system is typically only screened for the presence of an overt hearing loss. Our previous work suggested that children suspected of APD (sAPD) have atypically sharp cochlear tuning when measured using stimulus frequency otoacoustic emission (SFOAE) group delay. In the present work, we extend our previous findings and test the hypothesis that cochlear tuning influences auditory brainstem response (ABR) latencies. Our hypothesis is based on filter theory, which suggests that a sharper filter will take longer to build-up and ring longer. We predicted that sharper cochlear filters in sAPD should result

in delayed ABR wave latencies and will be associated with poorer performance on speech perception tests. Preliminary data from 16 sAPD and 6 typically developing children show a positive correlation between cochlear tuning and ABR peak I latency. Cochlear tuning explains a significant proportion of the variance in ABR peak I latency ($R^2 = 0.25$). Further behavioral results and implications for inclusion of auditory peripheral examination in APD tests will be discussed.

9:30–9:45 Break

9:45

3aPPa6. Transient developmental hearing loss leads to disruption of striatal function in adults. Todd M. Mowery (CNS, New York Univ., 4 Washington Pl., 1008, New York, NY 10003, tm106@nyu.edu)

Transient hearing loss during development can lead to deficits in the auditory perceptual abilities of children. These deficits arise from both peripheral ear and central neural dysfunction along the auditory neuraxis. Even if hearing loss resolves or is treated, persistent learning deficits and language impairments can occur. This suggests that regions downstream of the primary auditory neuraxis can be disrupted by brief auditory deprivation. One such region, which is highly involved in language development, is the auditory striatum. Thus I asked how the development of cellular properties in the auditory striatum are affected by hearing loss, and how these changes affect the learning of an auditory task. I used the gerbil and a brain slice preparation to assess how the expression of long term potentiation (LTP), which is a biomarker of learning, correlated with behavioral performance in adult animals that had undergone developmental hearing loss. I found that the cellular expression of LTP corresponded with the rapid acquisition of the behavioral task, and that developmental hearing loss delayed both the expression of LTP and the correlated behavioral acquisition. This suggests that striatal dysfunction could play a role in language acquisition impairments in children with hearing loss.

10:05

3aPPa7. Understanding pitch perception through physiological, modeling, and behavioural methods. Kerry M. Walker (Physiol., Anatomy & Genetics, Univ. of Oxford, Sherrington Bldg., Parks Rd., Oxford OX1 3PT, United Kingdom, kerry.walker@dpag.ox.ac.uk)

Pitch is a salient perceptual quality that underlies our musical experience, interpretation of speech, and ability to attend to one of multiple speakers. Yet the brain mechanisms that support this key percept remain unclear. We have measured the pitch discrimination of humans, ferrets, and mice on a 2-alternative forced choice task. These experiments show that species differ in their weighting of harmonic and temporal envelope cues for pitch judgments. By applying existing computational models, we demonstrate how cochlear filter properties can explain some of these species differences. We have also used microelectrode recordings in behaving ferrets to understand how individual cortical neurons represent pitch. We find that: (1) cortical activity correlates better with ferrets' pitch judgements than stimulus f0; and (2) inhibition can shape neural representations of f0 when animals are actively engaged in a pitch discrimination task, compared to when they are passively listening to the same sounds. We are now using 2-photon calcium imaging, during which we can measure the precise spatial position and responses of large numbers of individual neurons simultaneously, to better understand how and where pitch is extracted within the cortical microcircuit.

Contributed Papers

10:25

3aPPa8. Dynamic allocation of listening effort when listening to speech. Matthew Winn (Speech & Hearing Sci., Univ. of Washington, 1417 NE 42nd St, Seattle, WA 98105, mwinn83@gmail.com)

Measuring listening effort is an essential part of understanding the communication ability of a person with hearing loss. In this study, we are using pupillometry to measure how listener effort changes over the course of an utterance, in cases where the listener has reason to allocate effort earlier or later in time. Listeners completed a sentence-recognition task where high- or low-context sentences were followed by various stimuli (noise/digit sequences/speech) or silence. In cases where listeners can "restore" misperceived words with the aid of context, we see significant late-occurring reductions in pupil size when a sentence is followed by silence. This alleviation of cognitive activity is shown to be systematically disrupted by sounds (noise or speech) that are played after the end of the sentence. Furthermore, we show evidence of reallocation of effort earlier in time in cases where that disruption is expected. This underscores a largely underappreciated challenge in auditory perception, which is not limited to the coding stimulus itself, but also planning and post-stimulus processing. This phenomenon might underlie individuals' difficulty in real-life scenarios where a single mistake cannot be corrected in time, and cascades forward to disrupt perception of later utterances.

10:40

3aPPa9. Using functional near-infrared spectroscopy to assess auditory attention. Min Zhang and Antje Ihlefeld (Biomedical Eng., New Jersey Inst. of Technol., 323 Martin Luther King Blvd, Fenster Hall, Rm. 645, Newark, NJ 07102, ihlefeld@njit.edu)

Speech intelligibility in background sound can improve when listeners attend to the target. Using functional magnetic resonance imaging, previous work demonstrates that auditory attention alters Blood Oxygenation Level Dependent (BOLD) signals in the lateral frontal cortex (LFCx; Michalka *et al.*, Neuron 2015), at least when directed to spatial cues. Using functional near-infrared spectroscopy (fNIRS), we recorded BOLD responses from LFCx in 54 normal-hearing participants, each performing one of three auditory experiments. The weights of general linear modeling (GLM) fits of the fNIRS recordings were compared using analysis of variance. In experiment 1, when listeners selectively attended to stimuli differing in pitch and space, both left and right LFCx showed significant activation ($p = 0.01$ and 0.02 for left versus right LFCx). In contrast, passive listening to speech-shaped noise did not cause significant changes in LFCx activation ($p = 0.3$ and p 0.1 for left versus right LFCx). Experiment 2 confirmed these results, revealing comparable fNIRS traces when attending to pitch alone, versus attending to space and pitch. Experiment 3 further strengthened these results by controlling for eye movements. Result suggests that fNIRS is a viable tool for assessing whether or not a listener deploys auditory attention to perform a task.

Session 3aPPb

Psychological and Physiological Acoustics: Auditory Neuroscience Prize Lecture

William Yost, Chair
Arizona State University, PO Box 870102, Tempe, AZ 85287

Chair's Introduction—11:00

Invited Paper

11:05

3aPPb1. Cortical mechanisms underlying perception in complex auditory scenes. Shihab Shamma (Elec. and Comput. Eng. & Inst. for Systems Res., Univ. of Maryland College Park, AV WIlliams Bldg, College Park, MD 20742, sas@umd.edu)

Humans and many animals can easily attend to one of multiple similar sounds, segregate it, and follow it selectively over time even in extremely noisy environments. I shall review the psychological and neural underpinnings of this perceptual feat, explaining how it fundamentally depends on the representation of sound in the auditory cortex. I shall then outline how the attributes of simultaneous sounds (pitch, timbre, and location) are disentangled into separate streams, and bound into unified sources, by the temporal coherence of the cortical responses they evoke and their rapid adaptive properties. Recent neurophysiological results in support of these ideas will be discussed, especially in animals that have been trained to segregate and attend to the speech stream of a target speaker in a mixture. Finally, these findings will be related to analogous tasks in other sensory systems (visual object segregation in crowded scenes), leading to biologically inspired computational algorithms that may perform these tasks with no prior information or training.

Session 3aSC

Speech Communication: Session in Memory of James J. Jenkins

Kanae Nishi, Cochair
Boys Town National Research Hospital, 555 N. 30th Street, Omaha, NE 68131

Terry L. Gottfried, Cochair
Psychology, Lawrence University, 711 E. Boldt Way, Appleton, WI 54911

Linda Polka, Cochair
School of Communication Sciences & Disorders, McGill University, 2001 McGill College Avenue, 8th floor, SCSD, Montreal, QC H3Z 1Z4, Canada

Chair's Introduction—8:40

Invited Papers

8:45

3aSC1. Jim Jenkins—Mentor extraordinaire. Patricia K. Kuhl (Inst. for Learning & Brain Sci., Univ. of Washington, Box 357920, Seattle, WA 98195, pkkuhl@u.washington.edu)

James J. Jenkins (J^3 to his students) was the quintessential mentor. As a graduate student at the University of Minnesota in the late 1960s in Speech Communication, Jim was a member of my advisory committee, and a strong advocate of my interests and pursuits in the area of language and the brain. Jim set up a research fellowship for me with Dr. Hildred Schuell, a clinical scientist he deeply respected and collaborated with on publications, who was Director of the Aphasia Section in the Neurology Service at the VA Hospital in Minneapolis. I began to study the effects of cerebrovascular accident (stroke) on language, and was thrilled to be there. Unfortunately, Dr. Schuell succumbed after a sudden onset of cancer 9 months after my fellowship began. I remember asking Jim, "now what?" J^3's reply, and the 3-month assignment he gave me, changed my life, setting me on a course that I remain on to this day. I was never able to thank J^3 enough for his sage advice. Jim's talent at mentoring students—his ability to understand where they might shine—is something that inspires me every day.

9:05

3aSC2. When infants encounter infant speech. Linda Polka (School of Commun. Sci. & Disord., McGill Univ., 2001 McGill College Ave., 8th Fl., Montreal, QC H3Z 1Z4, Canada, linda.polka@mcgill.ca)

Over the past four decades, we have learned a great deal about how infants perceive and decode the speech spoken around them and directed to them. Yet we know very little about how infants perceive their own vocalizations or speech with the unique vocal properties of an infant talker. This leaves a serious gap in our understanding of infant language development. In this talk, I will present findings from a new line of research that begins to address this neglected aspect of infant speech development by exploring how infants perceive speech with infant vocal properties. The findings suggest that access to infant speech has a broad and significant impact—influencing receptive, expressive, and motivational aspects of speech development. I will highlight some of the valuable lessons I learned under the mentorship of James J. Jenkins that have steered me along this rich and exciting line of research.

9:25

3aSC3. Recent findings on children's lexical tone development: Implications for models of speech development. Puisan Wong (Speech and Hearing Sci., The Univ. of Hong Kong, Rm. 757 Meng Wah Complex, Pokfulam NA, Hong Kong, pswResearch@gmail.com)

Lexical tone is important for a majority of the world's languages. Prior research consistently reported that children mastered lexical tone production between 1;6 to 2;6 of age, supporting the prevailing notion of theories of speech acquisition that children acquire suprasegmental features early and rapidly before mastering the full set of segmental features. However, intriguingly, studies that examined children's Cantonese tone perception found that children failed to accurately identify Cantonese tones in monosyllabic words until after eight years old, five years after children's full mastery of tone production at three years old, which questioned early acquisition of suprasegmental features and the widely accepted assumption in speech acquisition models that speech perception precedes speech production. **Wong, Schwartz, and Jenkins (2005)** started a series of studies using a new paradigm that controlled lexical expectation in tone rating to examine lexical tone development in children and reported strikingly different results. Monosyllabic and disyllabic tones produced by five and six years old Mandarin-speaking children growing up in the U.S. and in Taiwan and Cantonese-speaking children growing up in Hong Kong were not adult-like. Tone production development was slower than tone perception development. The findings shed new light on universal models and theories of speech acquisition.

9:45

3aSC4. Voice quality: Speaker identification across age, gender, and ethnicity. Sonja Trent-Brown (Psych. Dept., Hope College, 35 E 12th St., Holland, MI 49423, trentbrown@hope.edu)

Listeners can perceptually identify speaker gender and ethnicity at better than chance guessing for adult speakers (Lass *et al.*, 1978; Thomas & Reaser, 2004; Trent-Brown *et al.*, 2012). These findings are supported across varying levels of phonetic complexity and across temporal manipulations that impact perceptual processing. There is less evidence regarding the extent to which these patterns hold for child speakers and whether listener gender/ethnic background and experience influence accuracy of identification. African American and European American male and female adult speakers were recorded producing passage-, sentence-, and word-length (/hVd/) stimuli including 11 General American English vowels. African American and European American male and female child speakers, ages 8–12, were recorded producing sentence and word stimuli. A temporal manipulation introduced to distort perceptual expectations presented stimuli to listeners in both forward and reversed conditions. Listeners were male and female African American and European American undergraduate students. Significant outcomes were observed for accuracy across speaker gender, speaker ethnicity, phonetic complexity, and temporal condition. In addition to accuracy of identification, listener confidence ratings, identification reaction times, and confidence reaction times also showed differential outcomes with respect to speaker age, gender, and ethnicity.

10:05–10:25 Break

10:25

3aSC5. "If a perceptual problem exists, what are its roots?" Types of cross-language similarity, hypercorrection, and unexplored root canals. Ocke-Schwen Bohn, Anne A. Ellegaard, and Camila L. Garibaldi (Dept. of English, Aarhus Univ., Aarhus DK-8000, Denmark, engosb@hum.au.dk)

James Jenkins had a keen interest in the nature of perceptual problems of nonnative listeners and in the perceived similarity of speech sounds. This presentation reports on a series of experiments which examined how well four different types of similarity predict nonnative speech perception. The different types are ecphoric and perceptual cross-language similarity, perceived within-language similarity, and acoustic similarity. One set of experiments examined how well cross-language perceptual assimilation (English to Danish) and within- language similarity ratings (English-English) of English consonants by Danish listeners predict Danes' identification of English consonants. Another set of experiments explored the roots of English listeners' discrimination problems for the closely spaced Danish unrounded front vowels by relating these problems to two types of perceived cross-language similarity (ecphoric and perceptual) and to acoustic similarity. Results suggest that each of the four types of similarity accounts for some of the perceptual problems, but none does so exhaustively, probably because the root system of these problems is affected by additional factors including nonnative listeners' hypercorrection and by perceptual asymmetries. The clear conclusion from these experiments is that Jim's question will keep us busy for quite some time. [Work supported by Carlsberg Foundation and *Inge Lehmanns Legat af 1983*.]

10:45

3aSC6. Tracking the time course of individuals' perception and production of coarticulated speech. Patrice S. Beddor (Dept. of Linguist, Univ. of Michigan, 440 Lorch Hall, 611 Tappan St., Ann Arbor, MI 48109, beddor@umich.edu)

Pioneering research by James J. Jenkins and colleagues has demonstrated the importance for speech perception of the dynamic information specified across the syllable. Collaborative work in our lab builds on this tradition of studying listeners' use of dynamic information by investigating perception of coarticulatory information as it unfolds in real time. We are currently studying listener-specific strategies for attending to coarticulatory dynamics, and are asking whether these strategies are linked to that individual's own production patterns. I will report the results of experiments that test the hypothesis that individuals who attend to coarticulatory information especially closely in perception also produce more consistent and extensive coarticulation. The perceptual measure is the time course of participants' use of coarticulated vowel nasality in CVNC words as measured via eye tracking; the production measure is the time course of these participants' nasal airflow while producing CVNC words. Results support our hypothesis: participants who produced earlier onset of coarticulatory nasalization were, as listeners, more efficient users of that information as it became available to them. Thus, a listener's use of the dynamics of speech is predicted, to some degree, by that individual's production patterns. [Work supported by NSF grant BCS-1348150.]

11:05

3aSC7. Catching a rabbit with a tetrahedron: A contextualist approach. Valeriy Shafiro (Commun. Disord. & Sci., Rush Univ. Medical Ctr., 600 S. Paulina Str., AAC 1012, Chicago, IL 60612, valeriy_shafiro@rush.edu)

A defining characteristic of Jim Jenkins' long and productive research career was his ability to reconcile contradictory experimental evidence within a coherent conceptual framework that leads to novel insights and experimental approaches. Grounded in the deep knowledge of history of psychology, Jim's involvement and expertise in different aspects of human behavior, including language, speech, memory, attention, visual, and auditory perception, resulted in important theoretical breakthroughs (and countless stimulating discussions). One such work is the tetrahedral model of memory experiments. It considers four large clusters of experimental variables: subjects, materials, scoring tasks, and testing procedures, whose mutual interactions determine experimental outcomes. [J.J. Jenkins "Four points to remember: A tetrahedral model of memory experiments," in *Levels of Processing in Human Memory*, ed. LS Cermak, FIM Craik, (1979)]. Initially proposed to account for inconsistent findings of memory experiments, the model's applications have expanded to other areas of perceptual training and education as a guide to maximize learning outcomes. Following Jim's lead, I will illustrate how the tetrahedral model can (1) improve the understanding of research findings from two ecologically significant sound classes, speech and environmental sounds, in normal hearing and cochlear implant listeners, and (2) guide the design of effective auditory training programs.

11:25–11:55 Panel Discussion

Session 3aSP

Signal Processing in Acoustics and Underwater Acoustics: Co-Prime Arrays and Other Sparse Arrays I

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

Kainam T. Wong, Cochair
Dept. of Electronic & Information Engineering, Hong Kong Polytechnic University, 11 Yuk Choi Road, Hung Hom KLN, Hong Kong

Invited Papers

9:00

3aSP1. Sampling with semi-coprime arrays. Kaushallya Adhikari (Louisiana Tech Univ., 600 Dan Reneau Dr, Ruston, LA 71270, adhikari@latech.edu)

This research introduces a new array geometry called Semi-Coprime Array (SCA) that has the potential to significantly increase the degrees of freedom of a Uniform Linear Array (ULA). An SCA interleaves three ULAs (Subarray 1, Subarray 2, and Subarray 3). Each SCA has two underlying coprime integers (M and N). Subarray 1 and Subarray 2 have undersampling factors based on the coprime integers M and N, while Subarray 3 is a full ULA with the number of sensors determined by the undersampling factors of Subarray 1 and Subarray 2. Interleaving the three subarrays results in a highly sparse non-uniform linear array. Taking the minimum of the three subarrays' outputs produces a pattern that is devoid of aliasing, yet offers the resolution of a full ULA with equal aperture. An SCA offers closed form expressions for sensor locations which many existing sparse arrays do not. Compared to other existing sparse arrays that offer closed form expressions for sensor locations, coprime arrays and nested arrays, an SCA saves more sensors and matches the peak side lobe height of a full ULA with less extension.

9:20

3aSP2. Investigations on n-tuple coprime arrays. Dane R. Bush and Ning Xiang (Architectural Acoust., Rensselaer Polytechnic Inst., 2609 15th St, Troy, NY 12180, danebush@gmail.com)

Coprime arrays so far combine two sparsely-spaced subarrays, undersampled by factors of M and N, in order to achieve MN degrees of freedom. To ensure that the grating lobes of each subarray can be largely eliminated, M and N must be coprime. In number theory, sets of pairwise coprime numbers can exceed just two numbers. The current work extends the theory to include coprime linear arrays with an arbitrary number, n, of subarrays. A triple coprime array comprised of $n = 3$ equal-aperture subarrays with M, N, and O elements, undersampled by factors of NO, MO, and MN, respectively, may use just $M + N + O - 1$ shifts to observe MNO directions. The design frequency of such an array not only exceeds the Nyquist spatial sampling limit, but is also greater than that of a standard (double) coprime array with equivalent aperture and number of elements. A triple coprime array is constructed with subarrays of $M = 3$, $N = 4$, and $O = 5$ and measured in a simulated free field condition. Experimental validation of the array confirms that the triple coprime array can also observe lower frequencies up to the design frequency. This paper discusses advantages and practical significance of n-tuple coprime microphone arrays over conventional double coprime linear arrays.

9:40

3aSP3. Effects of lag redundancy on array beam patterns and redundancy patterns for extended co-prime arrays. Andrew T. Pyzdek (Graduate Program in Acoust., The Penn State Univ., PO Box 30, State College, PA 16804, atp5120@psu.edu) and David C. Swanson (Appl. Res. Lab., Penn State Univ., State College, PA)

Lag redundancy describes the number of times a given spatial lag is measured by an array. For sparse arrays, the reduction of lag redundancies is generally desirable for lowering cost and processing overhead. However, as lag redundancies determine the natural weighting function of an array's spectral estimates, the pattern of redundancies dictates the natural beam pattern of a given array. We consider the impact of array sparsity on sidelobe level and array gain in both weighted and unweighted conventional beamforming. This analysis is then applied to co-prime arrays, for which redundancy patterns can be determined based solely on the choice of array length and co-prime factors. It is also shown that the spatial positions of lag redundancies are semi-regular, a desirable feature for sampling some complex environments.

10:00

3aSP4. Time difference of arrival-based acoustic source localization and separation technique for sparse arrays. Mingsian R. Bai (Power Mech. Eng., National Tsing Hua Univ., No. 101, Section 2, Kuang-Fu Rd., Hsinchu, Taiwan 30013, Taiwan, msbai63@gmail.com)

An acoustic source localization and separation technique intended for sparse arrays is suggested in this paper. Unlike conventional beamforming approaches, the proposed technique is based on time difference of arrival (TDOA) information. This effectively eliminates the spatial aliasing problem due to overly large inter-element spacing of sensors. This method begins with estimation of the TDOAs between at least 4 sensors for 3-dimensional sound fields, with the aid of the generalized crosscorrelation-phase transformation (GCC-PHAT) algorithm and the subspace-based time delay estimation algorithm. Next, a constrained least squares (CLS) algorithm is applied to locate the source based on the above-estimated TDOAs. Once the source location is found, the source signal is extracted by using the Tikhonov regularization (TIKR) algorithm and minimum variance distortionless response (MVDR) beamformer. Regularization parameter is selected with a special search procedure to best preserve the audio quality in the inverse solution process. Experiments and listening tests are undertaken to validate the proposed TDOA-based localization and separation technique.

10:20–10:35 Break

10:35

3aSP5. Applying concepts and tools from signal processing on graphs (SPG) to problems in array signal processing. Eldridge Alcantara, Les Atlas (Elec. Eng., Univ. of Washington, Seattle, Box 354090, Seattle, WA 98195, eealcant@uw.edu), and Shima Abadi (Mech. Eng., Univ. of Washington, Bothell, Bothell, WA)

Signal processing on graphs (SPG) is an emerging area of research that extends well-established data analysis concepts and tools to support a special type of signal where data samples are defined on the vertices of a graph. Since SPG emerged in 2013, fundamental operations such as filtering, the Fourier transform, and modulation have been formally defined that uniquely consider and take advantage of the underlying complex and irregular relationship between data elements which is captured mathematically by a graph. The purpose of this study is to analyze the applicability of SPG to array signal processing. We show that signals defined on a graph, or graph signals for short, are natural models for data collected over a line array of sensors. We also apply existing SPG processing algorithms to array signal data and investigate and probe whether SPG can help increase array gain.

10:55

3aSP6. Frequency-wavenumber analysis with a sparse array. Yonghwa Choi, Donghyeon Kim, and Jea Soo kim (Korea Maritime and Ocean Univ., 727 Taejong-ro, Youngdo-Gu, Busan 49112, Busan KS012, South Korea, hwa1470@naver.com)

In underwater acoustics, there has been many studies for finding the target direction using beamforming technique. When receiving a signal with a frequency higher than the design frequency of the array, it is difficult to estimate the direction of the signal due to spatial aliasing. In this study, we propose a method of estimating the direction of the frequency signal higher than the design frequency of the array by using frequency-wavenumber analysis. When the frequency-wavenumber analysis is performed, the striation pattern appears, and it is confirmed that the slope of the striation remains constant even if the spatial aliasing occurs. The direction of the signal was estimated by visual inspection and it was verified with SAVEX15 data.

11:15

3aSP7. Frequency-difference wavenumber analysis with a sparse array. Donghyeon Kim (Dept. of Convergence Study on the Ocean Sci. and Technol., KIOST-KMOU Ocean Sci. and Technol. School, 727 Taejong-ro, Yeongdo-Gu, 253, Ocean Sci. and Technol., Korea Maritime and Ocean University, Busan ASIKRKS012BUSAN, South Korea, donghyeonkim@kmou.ac.kr), Yonghwa Choi, Seongil Cho (Korea Maritime and Ocean Univ., Busan, South Korea), gihoon byun (Dept. of Convergence Study on the Ocean Sci. and Technol., KIOST-KMOU Ocean Sci. and Technol. School, Busan, N/A, Korea (the Republic of)), and Jea Soo kim (Korea Maritime and Ocean Univ., Busan, South Korea)

Frequency (f)- wavenumber (k) analysis can be used to estimate the direction of arrival (DOA) [J. Acoust. Soc. Am. **69**, 732–737 (1980)]. When the receiver is a sparse array that is not suitable for conventional plane-wave beamforming, it adversely causes aliasing error due to spatial sampling, thus many striation patterns can emerge in f-k domain. In this study, we propose frequency-difference wavenumber analysis that is motivated frequency-difference beamforming [J. Acoust. Soc. Am. **132**, 3018–3029 (2012)]. It is found that this approach can mitigate (or eliminate) such aliasing effect, which extends its applicability to the robust DOA estimation. Numerical simulation and experimental results are presented, and a major drawback is discussed.

Session 3aUW

Underwater Acoustics: Target Scattering in Underwater Acoustics: Imaging, Spectral Domain, and Other Representations I

Daniel Plotnick, Cochair
Applied Physics Lab., Univ. of Washington, Seattle, WA 98105

Timothy Marston, Cochair
APL-UW, 1013 NE 40th Street, Seattle, WA 98105

Chair's Introduction—8:00

Contributed Paper

8:05

3aUW1. Numerical determination of Green's functions for far field scattering solutions. Aaron M. Gunderson and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78758, aaron.gunderson01@gmail.com)

Typical finite element target scattering results are evaluated over a small physical domain and then propagated to the far field using the Helmholtz-Kirchhoff integral. This process allows for direct far field data-model comparison and is substantially faster than evaluation over a large domain. It does, however, require knowledge of the Green's function of the target's environment. For underwater buried targets, the two-medium Green's function pertaining to the water and sediment is often not readily known and can be cumbersome to analytically solve or approximate, particularly when surface roughness or volume inhomogeneity is to be incorporated. Instead, the Green's function can be determined directly through numerical methods. A direct process for numerical Green's function determination in an arbitrary two-medium environment is proposed, and is used to determine the far field scattering from buried targets. Results are compared to experimental scattering records on buried elastic targets, and to other models where the Green's function was determined/approximated analytically. [Work supported by Applied Research Laboratories IR&D and ONR, Ocean Acoustics.]

Invited Paper

8:20

3aUW2. Quantitative ray methods for scattering by tilted cylinders. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

After a review of ray methods and phenomena associated with high frequency scattering by spheres and cylinders in water viewed broadside, generalizations to tilted cylinders relevant to spectral and imaging domains will be summarized. These extensions were found to be useful for meridional as well as helical ray backscattering enhancements associated with leaky (or supersonic) waves on shells [F. J. Blonigen and P. L. Marston, J. Acoust. Soc. Am. **112**, 528–536 (2002)]. For such enhancements, Fermat's principle is useful for identifying ray paths of interest. In the case of helical waves (and in the broadside special case) the scattering amplitude can be expressed in terms of a Fresnel patch area where the guided wave is excited on the shell. Fresnel patches also give insight into the relatively large magnitude of meridional ray contributions. The coupling coefficient is proportional to the radiation damping of the leaky wave. For some shells it is necessary to take into account the anisotropy of the phase velocity. Computational benchmarks include scattering into the meridional plane by tilted infinite cylinders. Related phenomena include time-frequency domain features and enhancements from waves crossing truncations and from subsonic and negative group velocity guided waves. [Research supported by ONR.]

Contributed Papers

8:35

3aUW3. Backscattering enhancement due to weakly damped, leaky, guided waves on cylinders and associated images. Timothy D. Daniel, Sterling M. Smith, and Philip L. Marston (Phys. and Astronomy Dept., WSU, Washington State Univ., Pullman, WA 99164, timothy.daniel@email.wsu.edu)

In this work, distinct but related targets were studied using a circular synthetic aperture sonar system. Backscattering data from a solid brass cylinder were recorded. The response of the target was significantly spread out in time because leaky guided waves were weakly radiation damped. They gave significant backscattering contributions for a range of tilt angles. The associated meridional and helical wave contributions are also obvious in the frequency domain. (For comparison helical contributions are relatively weak for aluminum cylinders.) The second target is a bi-metallic cylinder made of a 1:1 brass cylinder bonded to a 3:1 aluminum cylinder. For a single-material cylinder, only 90 degrees of data are necessary due to symmetry. This target breaks that symmetry and requires a full 180 degrees of rotation. The distinction between the brass and aluminum ends is discernible in both the time and frequency domain. The join between the two metals affects the timing of certain guided waves that no longer travel the whole length of the compound cylinder. Image reconstruction was performed using Fourier based algorithms for both targets. The location of the compound cylinder joint is deducible from the location of features in the image domain. [Research supported by ONR.]

8:50

3aUW4. Separation of multiple backscattered echoes using dictionary updating sparse method. Xiangxia Meng, Xiukun Li (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145, Nantong St., Harbin, Heilongjiang 150001, China, mengxiangxia@hrbeu.edu.cn), and Andreas Jakobsson (Lund Univ., Lund, Sweden)

Sparse signal representation method has been proven to be effective to estimate and separate the multiple backscattered echoes from underwater target. Usually, the transmitted signal is used as a template to generate the dictionary. It is efficient under assumptions that the knowledge of signal components is perfectly known, or the deviations from the dictionary can be ignored. However, the properties of the echoes differ from the clean reference signal because of the scattering mechanism. In contrast, it has necessity to extract these deviations to have deep insight of the target. Herein, we propose a sparse method that allows the dictionary to be updated. The initial dictionary is defined as a usual way, i.e., based on the transmitted signal. Then, we use a constrained optimization problem to generate dictionary that differs from the original one, introducing an updating procedure. We introduce two constraints, one is to penalize the updated dictionary to be sparse and the other one is to ensure the elements do not deviate too much from the original dictionary. In this way, the resulting dictionary can be used to reconstruct the backscattered echoes, which is illustrated to be efficient with high accuracy by simulated and experimental results.

9:05

3aUW5. Experimental study on characteristics of echoes reflected by a cylindrical object in underwater multi-path channel. Fangyong Wang, Shuanping Du, and Jiao Su (National Key Lab. of Sci. and Technol. on Sonar, Hangzhou Appl. Acoust. Res. Inst., No. 715 Pingfeng Rd., LiuXia St., Hangzhou, Zhejiang 310023, China, sklwfy@yahoo.com)

Research work on characteristics of echoes scattered by a cylindrical object located in underwater multi-path is introduced in the paper. A sound scattering experiment of a cylindrical object in water-filled tank is carried out, the purpose of which is to give answers to questions of how the multi-path channel corrupt the characteristics of echoes from an elastic body in underwater multi-path channel and how many target-related features can still be extracted in both time and frequency domain which can be used for classification or feature-based detection. Detailed data processing results are shown and analyzed, which may be useful to research on signal processing method of sonar target detection and classification.

9:20

3aUW6. Coherent fusion of multi-aspect scans of UXO targets from CLUTTEREX16. Timothy Marston (APL-UW, 1013 NE 40th St., Seattle, WA 98105, marston@apl.washington.edu)

During the ONR and SERDP sponsored 2016 CLUTTEREX experiments, a large number of UXO were placed in a target field. These UXO were scanned by a sonar system mounted on a linear rail. Following each scan, the UXO were rotated in place by increments of 20 degrees to build up a multi-aspect acoustic characterization of each target. An automated process was subsequently developed to coherently fuse the results of these rotations to form 180-degree wide acoustic characterizations that could be used to represent the target in the complex image, frequency vs. aspect, or time vs. aspect domains. This talk will discuss the fusion process and the results for UXO and clutter objects in the target field.

9:35–9:50 Break

Invited Papers

9:50

3aUW7. Use of time, space, frequency, and angle data products in understanding target-in-the-environment scattering physics. Kevin Williams, Steven G. Kargl, and Aubrey L. Espana (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, williams@apl.washington.edu)

Broadband, multi-aspect time domain data of scattering from targets placed on sediment interfaces contains a wealth of information about the target and the modifications to the target scattering due to its location in the ocean environment. Different data products derived from the data can be used together to gain a better insight into the overall physical processes at play. We use a 2:1 aluminum cylinder and a 2:1 aluminum pipe to demonstrate the utility of different types of data products. Domains examined include imaging in the horizontal plane, frequency response as a function of look angle, and time domain response as a function of look angle. We then examine these same domains using relatively narrow windows of time with varying start times. This analysis allows visualization of the evolution of the physical mechanisms involved. [Sponsored by the Office of Naval Research and the Strategic Environmental Research and Development Program.]

10:05

3aUW8. Representation trade-offs for the quality assessment of acoustic color signatures. J. D. Park (The Penn State Univ. Appl. Res. Lab., P.O.Box 30, State College, PA 16804, jdanielpark@psu.edu), Daniel Cook (Georgia Tech Res. Inst., Smyrna, GA), and Alan J. Hunter (Univ. of Bath, Bath, United Kingdom)

Acoustic color is a representation of the spectral response over aspect, typically in 2-D. This representation aims to characterize the structural acoustic phenomena associated with an object with frequency and aspect as the two axes of choice. However, the strongest acoustic color signatures are typically the geometric features of an object due to geometric scattering. Naturally, these geometric features are well-represented as straight-line signatures in the spatial spectrum representation with the axes horizontal and vertical wavenumbers. Elastic responses due to other scattering mechanisms such as Rayleigh scattering or Mie scattering are typically smaller in magnitude, and are not easily recognizable or separable from the stronger responses. This work explores variants of acoustic color, to find intuitive representations for these types of scattering responses, and to assess their signature quality.

10:20

3aUW9. Elastic features in time-frequency representations for target classification. Rodolfo Arrieta, David E. Malphurs, Iris Paustian (Naval Surface Warfare Ctr. Panama City Div., 110 Vernon Ave., Panama City, FL 32407, rodolfo.arrieta@navy.mil), and Raymond Lim (Naval Surface Warfare Ctr. Panama City Div., Panama City, FL, FL)

Time-frequency representations aid in the interpretation of backscattered chirps, allowing a physical basis for selection of features for target classification using machine learning approaches. However, trade-offs in resolution, computational efficiency, noise reduction, and cross term rejection are made in these 2-D representations. Since our goal is the isolation of physically-meaningful features that will aid in the classification of targets, we are mainly concerned with the ability of these representations to reject noise. Towards this end, we have developed synthetic signals designed to robustly enhance the energy in the elastic portion of the acoustic backscattered return; thus, increasing the SNR for difficult targets. Previously, we demonstrated a capability to class-separate a diverse group of targets using features derived from time-frequency representations of signals collected at high SNR target aspects. In recent work this capability is being extended to other target aspects where SNR is weak by using our synthetic signals. This is demonstrated with cylindrical targets, where physics-based target models helped select the most robust elastic modes to use in the design of these synthetic signals. Results based on data collected from underwater unexploded ordnance and simple cylindrical targets will be presented. [This work was sponsored by SERDP and NSWC PCD.]

10:35

3aUW10. Utilization of aspect angle information in synthetic aperture images. Daniel Plotnick (Univ. of Washington, Applied Physics Lab., Seattle, WA 98105, plotnd@apl.uw.edu) and Timothy Marston (Univ. of Washington, Seattle, WA)

Synthetic aperture sonar (SAS) involves the creation of high resolution images of a scene via scattered signals recorded at different locations. Each pixel of the reconstructed image includes information obtained from multiple aspects due to the changing position of the sources/receivers. The aspect dependent scattering at each pixel may be exploited to provide additional information about the scene; we will present a framework for converting and utilizing multi-aspect data, as well as several examples. This aspect dependency may be leveraged to separate objects of interest from the background, to understand the local bathymetry, or for visualizing acoustic shadowing in circular aperture sonar (CSAS) images. Additionally, the aspect dependence of low-frequency elastic scattering from objects may be used to understand underlying scattering physics; a laboratory example is presented.

Contributed Paper

10:50

3aUW11. Laboratory simulation of backscattering by a hemisphere in a boundary wave-guide with an external source. Auberry R. Fortuner and Philip L. Marston (Phys. and Astronomy Dept., WSU, Pullman, WA 99164-2814, auberry.fortuner@wsu.edu)

While there have been various investigations of backscattering when a source and a target are situated in the same wave-guide, the situation when the source is external to the wave-guide is less well explored. For example, a target buried in a mud layer over sand [K. L. Williams, J. Acoust. Soc. Am. 140, EL504 (2016)], the mud layer may act as a wave-guide. The target signatures when the source is in the water column above the mud are modified. A laboratory experiment has been performed to isolate wave-guide effects on target backscattering, using a simplified wave-guide in water. The wave-guide is defined between a free water surface and a thin partially transparent acrylic plate placed below the water surface. A soft spherical target is half-exposed at the free-surface and source-receiver is below the plate. A geometric method-of-images model gives the timing arrivals of wave-guide echoes from an infinite number of back-scattering paths that undergo multiple reflections in the wave-guide and gives the frequency response of the wave-guide by summation of the contributions. This model gives a closed-form result in the far-field limit, and is compared to the experimental results in the time and frequency domains. [Work supported by ONR.]

11:05

3aUW12. Synthetic aperture sonar speckle noise reduction performance evaluation. Marsal A. Bruna, David J. Pate, and Daniel Cook (Georgia Tech Res. Inst., 7220 Richardson Rd., Smyrna, GA 30080, marsal.bruna@gtri.gatech.edu)

SAS (Synthetic Aperture Sonar) imagery always contains speckle, which is often thought of as a kind of multiplicative noise with respect to the underlying scene reflectivity. Speckle arises from the coherent interference of waves backscattered by rough surfaces within a resolution cell. Numerous image processing algorithms have been proposed to reduce speckle while preserving image features. Frequently, these algorithms are evaluated using actual SAS imagery; where the underlying noise-free image is unknown. The lack of noise-free images along with a limited number of images can lead to an incomplete estimation of algorithm performance. The use of various image quality and speckle reduction metrics is also required to accurately assess performance. In this paper, a unified framework using both simulated and real imagery is used to analyze the performance of prominent algorithms, such as multilook and anisotropic diffusion methods.

11:20

3aUW13. Simulating an object response with synthetic aperture sonar inverse imaging. James Prater (NSWC PCD, 110 Vernon Ave., Panama City, FL 32407, james.l.prater@navy.mil)

Synthetic aperture sonar (SAS) algorithms were developed in order to produce high resolution imagery from wide-beam wideband data. The high data rates associated with SAS data have fostered the development and implementation of efficient beamforming algorithms, many of which were originally developed for synthetic aperture radar. Some of these beamforming algorithms are invertible, and inverse beamforming techniques have been developed for SAS data analysis, where imagery is inverted in order to produce aspect-dependent data. Since imagery is invertible, this process can also be used as a simulation to produce stave-level data from a simulated image. This paper describes a modeling approach where inverse imaging techniques are applied in order to generate simulated stave-level sonar data. Using this approach, an image is generated from assumed object geometry and reflectance and inverted in order to generate stave-level data. The stave level data is then beamformed specifically to produce frequency vs. aspect data products. Since the inverse imaging process assumes specular reflection, features present in the frequency vs. aspect data are solely due to the surface reflectance and object shape. Comparison of inverse image simulated data with physics based simulations or field-collected data are shown and the limitations of the comparison are discussed.

11:35

3aUW14. Characterization of internal waves in synthetic aperture sonar imagery via ray tracing. David J. Pate, Daniel Cook (Georgia Tech Res. Inst., 7220 Richardson Rd., Smyrna, GA 30080, david.pate@gtri.gatech.edu), Anthony P. Lyons (Univ. of New Hampshire, Durham, NH), and Roy E. Hansen (Norwegian Defence Res. Establishment, Kjeller, Norway)

Synthetic aperture sonar imagery often captures features that appear similar to sand waves but are actually pockets of denser water traveling as isolated waves along the seafloor. These pockets of cold water refract acoustic waves like a lens, causing intensity peaks and shadows that resemble medium to large scale sand waves. This work uses dynamic ray tracing to predict the intensity return as affected by refraction. First, we explore the nature of the intensity pattern created by internal waves of various shapes and sizes. Then, we use an optimization-based approach to solve the inverse problem: given an intensity pattern, determine the size, shape, and location of the internal wave that created it.

Session 3pAA

Architectural Acoustics: AIA CEU Course Presenters Training Session

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

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 30 Lafayette Square - Suite 103, Vernon, CT 06066

Chair's Introduction—1:00

Invited Papers

1:05

3pAA1. Architectural acoustics short course presentation material. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

The Technical Committee on Architectural Acoustics (TCAA) is a Registered Provider in the American Institute of Architects (AIA) Continuing Education System (CES). The TCAA has developed a standardized introductory short course for architects, called "Architectural Acoustics." An architect can earn one continuing education unit (CEU) by attending this short course, if it is presented by a qualified member of TCAA. The course covers topics in sound isolation, mechanical system noise control, finish treatments, and implementation of quality acoustical spaces. This paper will cover the course material in order to prepare and qualify potential presenters. In order to qualify as an authorized presenter for this AIA/CES short course, attendance at this special session and membership in TCAA are required.

2:05

3pAA2. Architectural acoustics continuing education course—Presenter registration and reporting requirements. Bennett M. Brooks (Brooks Acoust. Corp., 30 Lafayette Square - Ste. 103, Vernon, CT 06066, bbrooks@brooks-acoustics.com)

The Technical Committee on Architectural Acoustics (TCAA) is a Registered Provider in the American Institute of Architects (AIA) Continuing Education System (CES). The TCAA has developed a standardized introductory short course for architects called "Architectural Acoustics," for which attendees can earn one continuing education unit (CEU). This paper will cover the administrative requirements of the AIA/CES, to prepare potential presenters. These requirements include the proper handling of paperwork so that AIA members may receive credit for the course. The manner in which the course is given is also dictated by AIA requirements. TCAA membership and attendance at this workshop are required to qualify as an authorized presenter for this AIA/CES short course. Of course, anyone is free to register with the AIA to provide their own CEU program. However, the advantages of participating in this program are that the TCAA short course is already prepared, it is pre-approved by the AIA, and the registration fees are paid by the Acoustical Society of America.

Session 3pAB

Animal Bioacoustics: Lessons on Auditory "Perception" from Exploring Insect Hearing

Norman Lee, Chair
Biology, St. Olaf College, 1520 St. Olaf Avenue, Northfield, MN 55057

Chair's Introduction—1:10

Invited Papers

1:15

3pAB1. Insect ear biomechanics: Passive and active processes that can inspire acoustic sensors. James F. Windmill (Dept. of Electron. & Elec. Eng., Univ. of Strathclyde, 204 George St., Glasgow G1 1XW, United Kingdom, james.windmill@strath.ac.uk)

Biological systems provide an incredible wealth of archetypes that have emerged through evolutionary processes. Hearing organs are a good example of how different solutions, and adaptions, across different animal taxa, can often converge to solve similar sensory problems. Hearing has evolved independently multiple times across the insects, and the diversity of these biological solutions therefore provides a wealth of inspiration for the creation of novel acoustic sensors. Some biological solutions can be considered as purely passive mechanical constructs that accomplish some processing of the incoming sound. This talk will consider examples of this, including the frequency discrimination of the locust ear, and the directionality and wideband response of different moth ears. Several insects also display active hearing processes, whereby energy is used to actively change the hearing response, bearing some similarity to the processes found in the mammalian inner ear. This talk will thus discuss how some moths actively tune their ear, and how the mosquito ear actively utilizes gain and compression. Finally, the talk will discuss some examples of different acoustic sensors from the University of Strathclyde that take inspiration from these passive and active biological processes.

1:35

3pAB2. Stay tuned: Active processes tune tree cricket ears to maintain a match to temperature-dependent song frequency. Natasha Mhatre, Gerald Pollack, and Andrew Mason (Dept. of Biological Sci., Univ. of Toronto, 1265 Military Trail, Mason Lab, Scarborough, ON M1C 1A4, Canada, natasha.mhatre@gmail.com)

Tree cricket males produce tonal songs, which are used to attract mates and to interact with other males. In other crickets, the mechanical properties of their peripheral auditory system have evolved to have resonant modes at the communication frequency. The tree cricket auditory system, however, is not passively tuned to song frequency. Tree crickets exploit an active amplification process to tune hearing to song frequency. However, their song frequency increases with temperature, presenting a problem for tuned listeners. We show here that the actively amplified frequency increases with temperature, and shifts mechanical and neuronal auditory tuning to maintain a match with changing song frequency. We also find that in tree crickets active amplification does not provide greater sensitivity or dynamic range than that observed in other crickets that lack active amplification. Thus, the primary adaptive function of active amplification is to ensure that auditory tuning remains matched to conspecific song frequency, despite changing environmental conditions and signal characteristics.

1:55

3pAB3. A neural circuit for song pattern recognition in the cricket brain. Berthold Hedwig (Zoology, Univ. of Cambridge, Dept. of Zoology, Downing St., Cambridge CB2 3EJ, United Kingdom, bh202@cam.ac.uk), Stefan Schoeneich (Inst. for Biology, Univ. of Leipzig, Leipzig, Germany), and Konstantinos Kostarakos (Inst. of Zoology, Karl Franzens Univ., Graz, Austria)

Acoustic communication is based on amplitude and frequency modulation of sound signals. Temporal features of the signal require processing by central auditory neurons, the brain circuits, however that detect temporal features are poorly understood. We show how five neurons in the brain of female field crickets form an auditory feature-detector circuit for the pulse pattern of the male calling song, and exhibit properties of a delay-line and coincidence-detection mechanism. The network receives its direct input from a single ascending auditory interneuron. An internal delay that matches the pulse period of the calling song is established by a non-spiking brain neuron. In response to a sound pulse, it generates a transient inhibition that triggers a delayed rebound depolarization. The direct input and the delayed responses converge in a coincidence detector neuron, which responds best to the pulse pattern of the species-specific calling song as the rebound activation of the non-spiking neuron coincides with the response of the ascending interneuron to the subsequent sound pulse. The output of the coincidence detector neuron is further processed by a feature detector neuron to suppress unselective responses and background activity. The circuit reveals principal mechanisms of sensory processing underlying the perception of temporal auditory patterns.

2:15

3pAB4. Stimulus specific adaptation in the auditory system of insects: Can a single neuron learn? Johannes Schul and Ryan Yost (Biological Sci., Univ. of Missouri, 207 Tucker Hall, Columbia, MO 65211, schulj@missouri.edu)

Stimulus-specific adaptation (SSA) is the suppression of a neuron's activity to repetitive stimuli while maintaining responsiveness to infrequent signals. In *Neoconocephalus* katydids, one auditory interneuron (TN-1) shows strong SSA in oddball paradigms, when standard and oddball pulses differ in carrier frequency. SSA occurred for pulse rates from >140 Hz down to 1 Hz. At fast repetition rates (>100 Hz), responses to the common pulses ceased, while oddballs elicit single spikes. At slower rates (<50 Hz), both standard and oddball pulses elicited spiking, with responses to oddballs being significantly larger, comparable to SSA described in vertebrate hearing systems. At slow rates, SSA also occurred when the shape of standard and oddball pulses differed, while having identical spectral properties. We identified at least two dendritic mechanisms that contribute to SSA at fast pulse rates (>100 Hz). The mechanisms underlying SSA at slow pulse rates are less well understood; likely, dendritic Ca^{2+}-gated currents contribute to SSA at slow pulse rates. At slow rates, response reduction to repeated pulses resembles habituation as oddball pulses cause dishabituation, i.e., the response to the following standard pulse is larger than that to the preceding one. Whether this habituation is generated within TN-1 or by its synaptic inputs remains an open question.

2:35

3pAB5. Making high-stakes decisions in complex acoustic environments: Revisiting the Cocktail Party problem in a multimodal sensory context. Daniel R. Howard (Biological Sci., Univ. of New Hamsphire, UNH Spaulding Hall G32, 38 Academic Way, Durham, NH 03824, daniel.howard@unh.edu), Norman Lee (Biology, St. Olaf College, Northfield, MN), and Carrie L. Hall (Biological Sci., Univ. of New Hamsphire, Durham, NH)

Ambient noise in its many forms represents an ecological reality and evolutionary driver that influences numerous expressions of animal behavior, especially those associated with communication. Noise often extends across sensory modalities, resulting in channel-specific effects that can range from non-additive to synergistic. Here, we describe how noise modality influences female preference for a preferred signal trait (low dominant frequency) in the lek-mating prairie mole cricket, *Gryllotalpa major*, a species whose mating system has evolved in the ecological context of both biotic and abiotic noise. We conducted two-choice playback experiments with females presented with male signals of the preferred vs. non-preferred trait in the context of isomodal (male chorus), cross-modal (substrate-borne vibration) and multimodal noise (both) conditions. We found that female preference for lower DF signals was robust to isomodal noise, but preference was weakened in treatments with a cross-modal stimulus only. Female performance in multimodal noise was comparable to control and isomodal treatments, suggesting that while cues obtained via the subgenual organs can mask salient airborne information, biotic noise may act as a releaser from this interference.

Session 3pBAa

Biomedical Acoustics: Biomedical Acoustics Best Student Paper Competition (Poster Session)

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

The ASA Technical Committee on Biomedical Acoustics offers a Best Student Paper Award to eligible students who are presenting at the meeting. Each student must defend a poster of her or his work during the student poster session. This defense will be evaluated by a group of judges from the Technical Committee on Biomedical Acoustics. Additionally, each student will give an oral presentation in a regular/special session. Up to three awards will be presented to the students with US $500 as first prize, US $300 as second prize, and US $200 as 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.

1pBA4. Wideband transskull refocusing of ultrasound beams using dual-mode ultrasound arrays: Ex vivo results
Student author: Hasan Aldiabat

1pBA7. Intranasal administration of temozolomide combined with focused ultrasound to enhance the survival of mice with glioma (A Pilot Study)
Student author: Dezhuang Ye

1pBA8. Broadband transskull multi-focus wavefront synthesis
Student author: Parker O'Brien

2aBA8.A nonlinear grid-search inversion for cortical bone thickness and ultrasonic velocities
Student author: Tho Tran

2pBA2. Estimating liver fat fraction from ultrasonic attenuation and backscatter coefficient measures in adults with nonalcoholic fatty liver disease
Student author: Lucia Albelda

2pBA11. Quantifying nonlinear elasticity modulus of tissue-like solids using acoustic radiation force
Student author: Danial Panahandeh-Shahraki

4pBAa1. In vivo ultrasound thermal ablation controlled using echo decorrelation imaging
Student author:Mohamed Abbass

4pBAa2. Patient-specific large-volume hyperthermia in the liver for ultrasound-enhanced drug delivery from thermosensitive carriers
Student author: Brian Chu

4pBAa6. Blood coagulation monitoring using acoustic levitation
Student author: Vahideh Ansari

Session 3pBAb

Biomedical Acoustics and Physical Acoustics: Induction Mechanisms for Bubble Nucleation II

Jeffrey B. Fowlkes, Cochair
Radiology, Univ. of Michigan, 3226C Medical Sciences Building I, 1301 Catherine Street, Ann Arbor, MI 48109-5667

Ronald Roy, Cochair
Engineering Science, University of Oxford, Parks Road, Oxford OX1 3PJ, United Kingdom

Invited Paper

1:40

3pBAb1. Tailoring cavitation nuclei for biomedical applications. Tyrone M. Porter (Boston Univ., 110 Cummington Mall, Boston, MA 02215, tmp@bu.edu)

Cavitation, or the creation and oscillation of bubbles, has a pivotal role in a variety of biomedical applications that involve ultrasound. For example, stable oscillating bubbles (stable cavitation) can reversibly permeabilize biological interfaces, including cell membranes and the blood-brain barrier. Bubbles that collapse inertially (inertial cavitation) radiate broadband emissions that can enhance ultrasound-mediated heating and thermal ablation. These cavitation-mediated bioeffects may provide medical benefit provided the location, type, and intensity of cavitation activity can be controlled. Microfluidic devices can be used to produce monodisperse microbubbles coated with viscoelastic shells that promote nonlinear oscillations. Liquid perfluorocarbon nanoemulsions can serve as long-circulating cavitation nuclei capable of extravasating into extravascular tissues. Using these technologies, cavitation nuclei can be tailored to provide a predictable acoustic response in a targeted location, thus providing a desired biological outcome.

Contributed Papers

2:00

3pBAb2. Correlation of cavitation activity with ultrasound-enhanced delivery of compounds to erythrocytes *ex vivo*. Emily M. Murphy, Mariah C. Priddy (BioEng., Univ. of Louisville, 2301 S Third St., Lutz Hall, Rm. 400, Louisville, KY 40292), Brett R. Janis, Michael A. Menze (Biology, Univ. of Louisville, Louisville, KY), and Jonathan A. Kopechek (BioEng., Univ. of Louisville, Louisville, KY, jakope01@louisville.edu)

Blood transfusions are one of the most common medical procedures in hospitals, but shortages of erythrocytes often occur due to their limited shelf life (6 weeks) when refrigerated. Preservation of erythrocytes in a dried state offers a potential solution to challenges faced with blood storage, and preservative compounds such as trehalose have been identified in organisms that survive desiccation in nature. However, these compounds do not readily cross mammalian cell membranes. Therefore, we are exploring the use of sonoporation to facilitate their delivery into erythrocytes *ex vivo*. In this study, we assessed the effect of cavitation activity on delivery of a fluorescent compound similar in size to trehalose (fluorescein) to human erythrocytes. Microbubbles were added to erythrocyte solutions and sonicated (2.5 MHz, 4 cycles) at various pressures and durations. Fluorescence was quantified with flow cytometry. The amplitude of broadband emissions in the first 8 seconds of sonication did not correlate with delivery ($r^2 = 0.23$), whereas after 8 seconds the broadband emissions amplitude was associated with increased delivery to erythrocytes ($r^2 = 0.97$). These results suggest that the timing of cavitation activity, rather than the amplitude alone, may be an important factor in ultrasound-mediated delivery of compounds into erythrocytes.

2:15

3pBAb3. Urinary stone erosion and fragmentation under low-intensity quasi-collimated ultrasound using gas-filled microbubbles with stone-targeting lipid shells. Yuri A. Pishchalnikov, William Behnke-Parks, Matt Mellema, Matt Hopcroft, Alice Luong (Applaud Medical, Inc., 953 Indiana St., San Francisco, CA 94107, yurapish@gmail.com), Tim Colonius, Kazuki Maeda (Dept. of Mech. and Civil Eng., California Inst. of Technol., Pasadena, CA), Kyle Morrison (Sonic Concepts, Inc., Bothell, WA), and Daniel Laser (Applaud Medical Inc., San Francisco, CA)

Urinary stone lithotripsy critically depends on the presence of cavitation nuclei at the stone surface. We hypothesized that introduction of stone-targeting microbubbles could increase cavitation activity at a stone surface sufficiently to allow stone erosion and fragmentation at peak negative pressures much lower than in acoustic energy-based urinary stone interventions with induced cavitation nuclei alone. Gas-filled microbubbles were produced with calcium-binding moieties incorporated into an encapsulating lipid shell. Stone surface coverage with these targeting microbubbles was found to approach an optimal (considering microbubble expansion during insonation) range of 5–15% with incubation times of three minutes or less. Using high-speed photomicroscopy, we observe bound microbubbles expanding 10- to 30-fold under insonation with quasi-collimated sources at mechanical indexes below 1.9. For observed stand-off parameters in the range of 0.2–0.6, the modeled collapse-generated shockwaves exceed 100 MPa. In swine model studies with these targeting microbubbles, stone fragmentation into passable fragments occurs with treatment times around 30 minutes, while post-treatment examination of ureters and kidneys shows no evidence of urothelium damage or renal parenchymal hemorrhage. The stone-targeting microbubbles reported on here have formed the basis for a new non-invasive urinary stone treatment which recently entered human clinical trials.

2:30

3pBAb4. Analytic prediction of histotripsy-induced bubble growth in an elastic medium. Kenneth B. Bader (Radiology, Univ. of Chicago, 5841 South Maryland Ave., MC 2026, Chicago, IL 60637, baderk@uchicago.edu)

Histotripsy is a form of therapeutic ultrasound that liquefies tissue mechanically via acoustic cavitation. Bubble growth due to histotripsy excitation has been calculated analytically with high accuracy in a fluid medium. Tissue elasticity is a determining factor in the therapeutic efficacy of histotripsy, and has not been considered in analytic bubble dynamics calculations. In this study, an analytic model to predict histotripsy-induced bubble expansion based on the medium surface tension, viscosity, and inertia was extended to include the effects of medium stiffness. Good agreement was observed between the predictions of the model and numerical computations. The predictions of the model were also consistent with experimental observation of bubble expansion, though are dependent on the elasticity form. Bubble growth was weakly dependent on the medium elasticity for highly nonlinear, shock scattering histotripsy pulses, but was strongly dependent for purely tensile, microtripsy pulses. For both forms of histotripsy, bubble growth was completely suppressed when the elastic modulus exceeded 20 MPa and the peak negative pressures was less than 50 MPa. These results highlight the importance of the histotripsy insonation scheme on bubble growth in an elastic medium, as well the range of tissue elasticities for efficacious bubble-induced liquefaction.

2:45

3pBAb5. On amplification of radial oscillations of microbubbles due to bubble-bubble interaction in polydisperse microbubble clusters under ultrasound excitation. Hossein Haghi, Amin Jafari Sojahrood, and Michael C. Kolios (Phys., Ryerson Univ., 350 Victoria St., Toronto, ON M5B 2k3, Canada, hossein.haghi@ryerson.ca)

Optimizing the performance of microbubbles (MBs) in applications not only requires a good understanding of the dynamics of individual MBs, but also knowledge of their interactions with each other within polydisperse clusters. However, the behavior of acoustically driven MB clusters is not well understood. Most studies have been limited to cases of a few interacting MBs for a limited range of acoustical parameters. We use a numerical method to simulate the dynamics of clusters of (20–50 MBs) randomly distributed MBs. MBs with the size distribution of 2–8 micron were sonicated with pressures between 1 and 250 kPa, frequencies between 0.5 and 10 MHz and concentrations between 2e3 and 2.5e6 MBs/mL. Bifurcation structures of radial oscillations of each MB within a cluster were studied. Results show that in a polydisperse MB cluster, larger MBs (even in small numbers) can amplify the radial oscillations of smaller MBs. Amplification occurs if the resonance peaks of larger MBs coincide with the resonance peaks of smaller MBs. Additionally, there is an optimum number density at which amplification is maximized. Results suggest that the presence of bigger MBs may enhance the superharmonic oscillations of the smaller MBs which can be applied in superhamonic ultrasound imaging using MBs.

WEDNESDAY AFTERNOON, 9 MAY 2018 — GREENWAY C/D, 2:15 P.M. TO 3:20 P.M.

Session 3pID

Interdisciplinary: Hot Topics in Acoustics

Christina J. Naify, Chair
Jet Propulsion Lab, 4800 Oak Grove Dr, MS 157-316, Pasadena, CA 91101

Chair's Introduction—2:15

Invited Papers

2:20

3pID3. An objective metric for describing the basic acoustics of binaural directivity patterns in humans. Andrew Dittberner, Changxue Ma (Res. Group, GN Adv. Sci., 2601 Patriot Blvd., Glenview, IL 60026, adittberner@gnresound.com), and Rob de Vries (Res. Group, GN Adv. Sci., Eindhoven, Netherlands)

It is accepted knowledge that having two ears are better than one when trying to listen to a signal of interest in the presence of spatially-separated noise sources (e.g. Blauert, 1997; Bregman, 1994; Zurek, 1993). Models have been proposed purporting of the benefits of the head shadow effect, binaural interactions, and cognitive factors that explain how one can understand sound with linguistic or other contextual meaning better in the presence of spatially-separate noise sources. However, less discussed is the attribute of human listeners having the ability to also hear and identify sound sources, seemingly on demand, that occur around them, a condition made possible by the fact of having two ears. Zurek (1993) proposed and discussed at length on the directivity effects of binaural listening (e.g., Better Ear Strategy). What is proposed in this study is an extension to this model to include the omni-directional directivity effects of binaural listening to describe the listener's ability to remain connected and aware of the sound landscape that surrounds them. Where the head shadow effect plays a role in improving the signal-to-noise ratio in one of the two ears, this strategy looks at how the two ears, due to their geometric location on the head, allows for the head to be acoustically transparent and keeps the listener connected to their surrounding sound landscape.

2:40

3pID2. Hot topics in structural acoustics and vibration: Advances in vibroacoustic modeling and novel materials. Robert M. Koch (Chief Technol. Office, Naval Undersea Warfare Ctr., Code 1176 Howell St., Bldg. 1346/4, Code 01CTO, Newport, RI 02841-1708, Robert.M.Koch@navy.mil)

Two major areas of current research focus in Structural Acoustics and Vibration (SAV) are in (1) the development of advancements in the Physics-Based Modeling (PBM) and simulation of real-world, large-scale vibroacoustic systems and also in (2) the development and incorporation of novel engineered materials to improve structural acoustic response performance from a broad array of perspectives. This paper presents examples from a wide survey of recent SAV computational PBM research, from the inclusion of additional complex, coupled modeled phenomena, to improvements in modeling efficiency, accuracy, and/or full frequency spectrum response prediction, through computational advances in the ability to solve extremely large-scale, previously intractable, structural acoustics models. Related to the second materials-based SAV research concentration, recent advancements and innovations in SAV-relevant materials including acoustic metamaterials, zero- and negative-Poisson's ratio materials, single crystal ceramics, nanomaterials, etc., are also presented.

3:00

3pID1. Model-based Bayesian signal processing in acoustics. Ning Xiang (School of Architecture, Rensselaer Polytechnic Inst., Greene Bldg., 110 8th St., Troy, NY 12180, xiangn@rpi.edu)

Bayesian signal processing has been increasingly applied to a wide variety of acoustical research and engineering tasks. Bayesian probability theory provides acousticians with an elegant framework for inferential data analysis which facilitates learning from acoustic experimental investigations that provide an improved understanding of the underlying theory. In these inferential analysis tasks, certain prior knowledge is often available about the acoustical phenomena under investigation, based either on the underlying physical theory or on certain phenomenological relationships. Bayesian probability theory allows this available information to be incorporated in the processing and analysis and exploited in the Bayesian framework as physical or phenomenological models. Many analysis tasks in acoustics often include two levels of inference, the model selection and the parameter estimation. Bayesian signal processing provides solutions to these two levels of inference by extensively using Bayes' theorem within this unified framework. This talk will discuss various model-based approaches recently applied to signal processing and analysis in acoustics using either one or both levels of inference.

WEDNESDAY AFTERNOON, 9 MAY 2018 GREENWAY J, 1:00 P.M. TO 3:05 P.M.

Session 3pPA

Physical Acoustics, Biomedical Acoustics and Psychological and Physiological Acoustics: Ultrasound and High Frequency Sound in Air in Public and Work Places: Applications, Devices, and Effects

Timothy Leighton, Cochair
Craig N. Dolder, Cochair
Institute of Sound and Vibration Research, University of Southampton, Highfield Campus, Southampton SO17 1BJ, United Kingdom

Chair's Introduction—1:00

Invited Papers

1:05

3pPA1. Ultrasound, human health, safe levels, and Cuba: What do we know? Timothy Leighton (Inst. of Sound and Vib. Res., Univ. of Southampton, Highfield, Southampton, Hampshire SO17 1BJ, United Kingdom, tgl@soton.ac.uk)

While definitive statements are impossible, a slim evidence base suggests the following on weight-of-probabilities regarding ultrasonic adverse effects in humans. One study at extremely high intensities reports physical effects (notably burning between fingers/nostrils). At lower intensities, adverse psychological effects occur in only a subset of the population (susceptibility possibly decreasing with age), are restricted to frequencies below ~25 kHz, probably result from the extraordinary sensitivity of our hearing/balance systems, and can be difficult to separate from (and may be causally related to) adverse effects of anxiety and annoyance. This does not remove the need for protection, especially for increasingly common public exposures, where the exposure and exposed person are often uncharacterised. Only one interim guideline from 1984 addresses maximum permissible levels (MPLs) for public exposure. It is based on scant evidence, and may or may not be appropriate. All other guidelines relate to occupational exposure. These MPLs are a legacy of decades of copying previous guidelines, which were themselves based on inadequate sampling (usually a small cohort of adult men), and averaging practices which obscured the particular sensitivities of a subset of the population. Against this background, the likelihood, or not, of an ultrasonic weapon in Cuba will be discussed.

1:25

3pPA2. Adverse effects of very high-frequency sound and ultrasound on humans. Mark D. Fletcher (Inst. of Sound and Vib. Res., Univ. of Southampton, 15 Consort Rd., Eastleigh So50 4JD, United Kingdom, mdf1f15@soton.ac.uk), Sian Lloyd Jones (Dept. of Audiol. and Hearing Therapy, Royal South Hants Hospital, Southampton, United Kingdom), Craig N. Dolder, Paul White, Timothy Leighton, and Ben Lineton (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, Hampshire, United Kingdom)

For many years, workers have reported adverse symptoms resulting from exposure to very high-frequency sound (VHFS) and ultrasound (US), including annoyance, dizziness and difficulty concentrating. Recent work showing the presence of a new generation of VHFS/US sources in public places has reopened the debate about whether adverse effects can be caused by exposure to VHFS/US. Our field measurements of VHF/US sources in public places have identified devices producing up to 100 dB SPL at 20 kHz. Nearly all of the sources measured, including those in places occupied by tens of millions of people each year, are likely to be clearly audible to many young people. We have conducted two studies. The first looked at adverse symptoms resulting from exposure to audible VHFS/US, and the second was a double-blind study of adverse symptoms resulting from exposure to inaudible VHFS/US. In each study, both symptomatic participants, who reported previously experiencing symptoms, and asymptomatics participants, who did not, were tested. We found evidence that symptoms were produced by exposure to audible VHFS/US but not by inaudible sound. It is possible that the substantial effects reported for inaudible VHFS/US exposure were not reproduced because of ethical restrictions on stimulus level and duration.

1:45

3pPA3. Measurement of ultrasound radiated from rodent repellents used in an occupational space, and auditory evaluation of the sound. Mari Ueda (Information Technol., Gent Univ., 4-9-1 Shiobaru, Fukuoka, Minami-ku 815-8540, Japan, m-ueda@design.kyushu-u.ac.jp)

In order to clarify the emission levels and hearing status of ultrasound used in an occupational spaces by pest control companies, three surveys were performed. The results of the acoustic field measurement showed that the rodent repellent device had a peak level and frequency of approximately 100 dB and 19 kHz, respectively. The results of an auditory evaluation experiment with 51 adult workers showed that younger workers recognized the ultrasound from the electronic rodent repellent device more clearly than the elderly workers.

2:05

3pPA4. Ultrasound measurements in the work environment. Jan Radosz and Dariusz Pleban (Central Inst. for Labour Protecion - National Res. Inst., Czerniakowska, 16, Warsaw 00-701, Poland, jarad@ciop.pl)

For the frequency range above 20 kHz, there is no clear and complete information on the factors influencing the result of a measurement of sound pressure level. Moreover, there are no current international standards for performing measurements of ultrasound at work stations. The authors presents a possibility for the adaptation of the existing measurement methods, in particular, the requirements for measuring instruments, procedures to be followed while performing measurements, the application of a correction to measurement results, and the determination of measurement uncertainty. The development of a consistent method of ultrasound measurement is of utmost importance in carrying out an assessment and reducing the risk of exposure to this physical factor in the work environment.

2:25

3pPA5. Low power wireless communication between personal electronic devices and hearing aids using high frequency audio and ultrasound. Jonathon Miegel (ARC Training Ctr. in Biodevices, Swinburne Univ. of Technol., 545 Burwood Rd, Hawthorn, VIC 3122, Australia, jmiegel@swin.edu.au), Philip Branch (Dept. of Telecommunications, Elec., Robotics and Biomedical Eng., Swinburne Univ. of Technol., Hawthorn, VIC, Australia), and Peter Blamey (Blamey Saunders hears, East Melbourne, VIC, Australia)

Hearing aids continue to be the main intervention for hearing loss but their ease of use and control is of concern due to their small size. While technological advances in Bluetooth Low Energy have allowed for improved wireless control, in particular, between personal electronic devices, its use for communication with hearing aids is problematic due to their limited battery life. This research investigates the implementation of acoustic wireless communication between personal electronic devices and hearing aids using multiple modulation schemes utilizing frequencies between 16 and 20 kHz. The performance of each modulation scheme is assessed over a 3 metre range and the power consumption compared to that of Bluetooth Low Energy.

2:45

3pPA6. A scavenger hunt using ultrasonic geocaches. Craig N. Dolder (Inst. of Sound and Vib. Res., Univ. of Southampton, Highfield Ave., Southampton, Hampshire SO17 1BJ, United Kingdom, dolder@utexas.edu)

Sound is commonly used for either communication or navigation. An ultrasonic scavenger hunt was designed that does both and is designed to raise awareness about acoustics. This scavenger hunts utilizes ultrasonic geocaches to both give information to the participants and educate them on topics including, the fact that sound may not be audible, the concept of hearing loss, other animals hear at different frequencies, and general facts from the hosting event. The geocaches use the frequency band above typical human hearing but still within the bandwidth of most personal electronics, 20 kHz–22 kHz. This band can be picked up by common smartphones and tablets and viewed using free spectrogram applications. The maximum sound pressure level output by the geocache devices falls below maximum public exposure recommendations but the signal is still visible on a spectrogram. The scavenger hunt was trialed at a science engagement event at the University of Southampton with over 6000 in attendance.

Session 3pPP

Psychological and Physiological Acoustics and Speech Communication: Acoustics Outreach to Budding Scientists: Planting Seeds for Future Clinical and Physiological Collaborations '18

Elin Roverud, Cochair
Dept. of Speech, Language & Hearing Sciences, Boston University, 635 Commonwealth Avenue, Boston, MA 02215

Anna C. Diedesch, Cochair
Communication Sciences & Disorders, Western Washington University, 516 High St., MS 9171, Bellingham, WA 98225

Chair's Introduction—1:10

Invited Papers

1:15

3pPP1. Auditory, cognitive, and linguistic processing skills in individuals with hearing loss. Shivali Appaiah Konganda (Dept. of Linguist, Macquarie Univ., Ground Fl., Australian Hearing Hub, 16 University Ave., Sydney, NSW 2109, Australia, shivali.appaiah-konganda@students.mq.edu.au), Mridula Sharma, Jessica J. Monaghan (Dept. of Linguist, Macquarie Univ., North Ryde, NSW, Australia), Gitte Keidser, Joaquin Tomas Valderrama Valenzuela (National Acoust. Labs., Australian Hearing, Australia, North Ryde, NSW, Australia), John Newall (Dept. of Linguist, Macquarie Univ., North Ryde, NSW, Australia), and Elizabeth Beach (National Acoust. Labs., Australian Hearing, Australia, North Ryde, NSW, Australia)

Hearing impairment affects a person's ability to communicate effectively. People with hearing loss (HL) report difficulty communicating in noise, even when the HL is compensated by conventional amplification. This study aims to investigate factors that contribute to understanding speech in noise. Nine adults with HL and nine controls participated in the study. The test-battery include auditory, cognitive and linguistic tests. For the HL group, auditory stimuli were filtered with NAL-RP prescription to compensate for their HL. Results indicate a significant difference in performance between the groups on the Modulation Detection Threshold (MDT) test [$F(1, 15) = 3.24$, $p = 0.04$] and the speech recognition in noise test [$f(1, 15) = 25.6$, $p<0.001$]. HL group performed better on the MDT and poorer at recognising speech in noise possibly due to broadening of auditory filters. With the broadened auditory filters in mind, this result supports the fact that they would have poor frequency specificity, detrimental for speech recognition. HL group performed better than the control group on the cognitive spare capacity test [$f(1, 15) = 4.72$, $p= 0.04$]. Preliminary data suggests that adults with HL may compensate for hearing-related difficulties in certain situations by using their cognitive skills.

1:35

3pPP2. Musical emotion recognition in bimodal patients. Kristen D'Onofrio (Vanderbilt Univ., 1215 21st Ave. S., Nashville, TN 37232, kristen.l.donofrio@vanderbilt.edu), Charles Limb, Meredith Caldwell (UCSF, San Francisco, CA), and René Gifford (Vanderbilt Univ., Nashville, TN)

Several cues are used to convey musical emotion, the two primary being musical mode and musical tempo. Specifically, major and minor modes are regarded as having positive and negative valence, respectively, and songs at fast tempi (i.e., quarter note = 92-196, Gosselin *et al.* (2005); quarter note = 80-255, Peretz *et al.* (1998)) are associated with more positive valence compared to songs at slow tempi (i.e., quarter note = 40-60, Gosselin *et al.* (2005); quarter note = 20-100, Peretz *et al.* (1998)). Caldwell *et al.* (2015) demonstrated that CI users relied on tempo (fast vs. slow) with little regard for mode (major vs. minor) when interpreting musical emotion, a finding consistent with spectral cues being poorly represented in CI users. The current study is an extension of Caldwell *et al.* (2015) examining how mode and tempo cues impact musical emotion recognition in bimodal listeners. Our primary hypothesis is that, unlike CI only users, bimodal listeners—with access to F0 and fine structure—will utilize both mode and tempo cues in a manner similar to normal-hearing listeners. Our secondary hypothesis is that low-frequency spectral resolution in the contralateral ear (via tuning curves and spectral modulation detection) will be significantly correlated with degree of bimodal benefit for musical emotion perception.

1:55

3pPP3. Perceptual and neural representation of consonants in hearing impaired listeners. Yaqing Su (Starkey Hearing Res. Ctr., 44 Cummington St, Boston, MA 02215, ysu11@bu.edu), Narayan Sankaran (Starkey Hearing Res. Ctr., Sydney, New South Wales, Australia), and Jayaganesh Swaminathan (Starkey Hearing Res. Ctr., Berkeley, CA)

Difficulties in perception of consonants have been associated with speech perception difficulties in hearing-impaired (HI) listeners. However, the neural bases underlying such difficulties have not been clearly explored and understood. The goal of this study was to use scalp electroencephalography (EEG) recordings to better understand the neural mechanisms that contribute to the consonant perception

difficulties in HI listeners. Such a psychophysiological approach can lead to the development of improved hearing-aid fitting and speech enhancement strategies. Perceptual and EEG responses to vowel-consonant-vowel speech were measured in 8 HI listeners with and without amplification. A machine-learning classifier was trained to discriminate the EEG signal evoked by each consonant. The performance of the classifier was compared to the HI listeners' psychophysical performance. For all subjects, consonant intelligibility was better in aided compared to unaided listening condition, but overall performance was well below ceiling. An information transmission analysis showed that place and manner of articulation were more affected than voicing and nasality. EEG waveform showed different response patterns for each consonant, and the machine-learning classifier was able to successfully "decode" consonants from the EEG signal. However, a straightforward relationship between the neural and perceptual representation of consonants could not be established in HI listeners.

2:15

3pPP4. Characterizing cortical auditory networks in bilateral cochlear implant users using electroencephalography. Daniel Smieja (Univ. of Toronto, The Hospital for Sick Children, 555 University Ave., Toronto, ON M5G1X8, Canada, daniel.smieja@mail.utoronto.ca), Benjamin Dunkley, Blake Papsin, and Karen Gordon (The Hospital for Sick Children, Toronto, ON, Canada)

Objective: This research aims to characterize the cortical network involved in listening with bilateral cochlear implants (CIs) through measures of functional connectivity. *Rationale*: A better understanding of the underlying cortical network involved in hearing will help elucidate remaining challenges experienced by children who received bilateral cochlear implants early relative to their peers with normal hearing. This work extends our previous focus on plasticity in temporal regions to include multiple cortical areas and considers the phase relationships between neural source signals. *Methods*: The data were collected using 64-channel electroencephalography in response to click stimuli in a passive listening condition. Source reconstruction was applied using the TRACS beamformer in order to localize the underlying neural generators. Sources were mapped to regions in the Automated Anatomical Labeling atlas and connectivity analyses were performed to examine statistical relationships between the atlas regions in order to generate network models. *Results*: Preliminary data show that in addition to the primary auditory areas, regions in the precuneus and frontal areas are activated in response to sound. The connectivity analyses are ongoing. *Significance*: Differences in the cortical auditory networks will help to understand how the auditory system is integrated in the cortex in bilateral cochlear implant users.

2:35

3pPP5. Tone-evoked acoustic change complex (ACC) in an animal model. Alessandro Presacco and John C. Middlebrooks (Otolaryngol., Univ. of California, Irvine, Medical Sci. D, Rm. D404, Irvine, CA 92697, presacca@uci.edu)

The auditory change complex (ACC) is a cortical evoked potential complex generated in response to a change (e.g., frequency or level) within an ongoing auditory stimulus. The ACC has been recorded in both normal-hearing human subjects and in cochlear implant users, suggesting that the ACC would be useful in clinical applications. Here, we investigate the feasibility of recording ACC in response to frequency or level changes in sedated cats. Five purpose-bred cats were sedated with ketamine and acepromazine. Continuous tones alternated between high and low frequencies or levels in 500-ms blocks. Frequency and level steps were varied parametrically. Scalp potentials were recorded with needle electrodes (two active electrodes = one on each hemisphere, reference: mastoid, ground = back of the cat). ACC was successfully elicited in all cats by both frequency and level steps. In many cases, ACCs were markedly greater for increasing or for decreasing stimulus steps. The discrimination thresholds measured were in good agreement with previous behavioral studies in which cats where trained to perform similar frequency or level discriminations. The results indicate that the ACC will be a useful tool for evaluating novel acoustical or electrical stimulation modes that are not yet feasible in humans.

2:55

3pPP6. Activity in human auditory cortex represents spatial separation between concurrent sounds. Martha Shiell, Lars Hausfeld, and Elia Formisano (Dept. of Cognit. Neurosci., Maastricht Univ., Oxfordlaan 55, Maastricht 6229 EV, Netherlands, marthashiell@gmail.com)

Auditory spatial information can enhance stream segregation in an auditory scene, but little is known about how auditory space is represented in the human cortex. If spatial cues are used for scene analysis, then it is the distance between sounds rather than their absolute positions that is essential, and thus, we hypothesized that auditory cortical neurons encode separation between sounds rather than absolute location. To test this hypothesis, we measured human brain activity in response to spatially-separated concurrent sounds with magnetic resonance imaging at 7 Tesla. Stimuli were spatialized amplitude-modulated broadband noises, recorded for each participant via in-ear microphones prior to scanning. Using a linear support vector machine classifier, we investigated if sound location and/or spatial separation between sounds could be decoded from the activity in Heschl's gyrus and the planum temporale. The classifier was successful only when comparing patterns associated with the conditions that had the largest difference in perceptual spatial separation, and not in conditions where only location changed and separation remained constant. Our pattern of results suggest that the representation of separation is not merely the combination of single locations, but is an independent feature of the auditory scene.

Session 3pSA

Structural Acoustics and Vibration, Signal Processing in Acoustics and Noise: Model Reduction for Structural Acoustics and Vibration

Kuangcheng Wu, Cochair
Naval Surface Warfare Center - Carderock, 9500 MacArthur Blvd, West Bethesda, MD 20817

Hubert S. Hall, Cochair
Mechanical Engineering, The Catholic University of America, 620 Michigan Ave. NE, Washington, DC 20064

Chair's Introduction—1:30

Invited Papers

1:35

3pSA1. Applications of the hybrid finite element method. Nickolas Vlahopoulos and Sungmin Lee (Univ. of Michigan, 2600 Draper Dr., Ann Arbor, MI 48109, nickvl@umich.edu)

The hybrid finite element analysis (Hybrid FEA) method is based on combining conventional finite element analysis (FEA) with energy finite element analysis (EFEA) for expediting the FEA computations when very dense models are needed. The difficulty in using conventional FEA at higher frequencies originates from requiring a very large number of elements in order to capture the flexible wavelength of the panel members which are present in a structure. In the Hybrid FEA the conventional FEA model is modified by de-activating the bending behavior of the flexible panels in the FEA computations and introducing instead a large number of dynamic impedance elements for representing the omitted bending behavior. The excitation is considered to be applied on the conventional FEA model and the vibration analysis is conducted. The power flow through the dynamic impedance elements is computed and applied as excitation to the EFEA model of the flexible panels. The EFEA analysis computes the vibration of the flexible panels. In the past, the Hybrid FEA has been utilized successfully for evaluating the vibration is production automotive and rotorcraft structures. A brief theoretical background will be reviewed, the practical aspects of the method will be discussed, and results from previous correlation studies will be presented.

1:55

3pSA2. A symmetrical formulation of coupled BEM/FEM for submerged structures based on acoustical reciprocity. Pei-Tai Chen (Dept. of System Eng. and Naval Architecture, National Taiwan Ocean Univ., No.2,Pei-Ning Rd. Keelung 20224, Keelung Keelung 20224, Taiwan, ptchen@mail.ntou.edu.tw)

The paper presents a structural acoustics formulation for elastic structures submerged in water. The structural equation is described by a finite element method where the linear displacement variables on the wetted surface are chosen locally as one normal displacement and the other two displacements tangent to the wetted surface, whereas the rest of rotational displacements or degrees of freedom not contacting with water are defined globally. A boundary element formulation describing the acoustic loading on the structure is expressed as a function of normal velocity or displacement on the wetted surface. The coupling of the FEM and BEM is through the normal velocity, or equivalently, the normal displacement on the wetted surface. An acoustical reciprocity is used to prove that the associated acoustic loading expressed as the normal displacement of the wetted structure is a complex symmetric matrix. This matrix can be viewed an acoustic element whose degrees of freedom are the normal displacements of the wetted surface. Thus, the coupled FEM and BEM becomes a symmetric banded matrix formulation, leading to an efficient numerical way to solve the equation. A capped cylindrical shell with periodical ring stiffeners and bulkheads submerged in water is used to demonstrate the present numerical method.

2:15

3pSA3. Finite element computation of the scattering response of objects buried in ocean sediments. Anthony L. Bonomo (Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd, West Bethesda, MD 20817, anthony.bonomo@gmail.com) and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

To assist in locating objects at or near the ocean floor, there is a need for computational models that can predict the scattering response of objects fully or partially buried in ocean sediments such as sand. In reality, sand can best be understood as a two-phase porous medium and the shear waves supported by the sediment can contribute meaningfully to the acoustic signature of the buried objects. Due to the slow phase speeds of these shear waves, the computational burden of finite element models can be excessive. In this talk, efficient modeling techniques and methods for reducing the computational time of these models are covered. [Work supported by ONR, Ocean Acoustics.]

2:35

3pSA4. A procedure for ship identification through acoustic signal processing. Gee-Pinn J. Too (Dept. of Systems and Naval Mechatronic Eng., National Cheng Kung Univ., No. 1 University Rd. NCKU, Tainan 70101, Taiwan, toojames@yahoo.com)

All ship's mechanical structures are different. Hence, the radiated noise spectra are different, especially in line spectra. Line spectra are composed of the spectra due to shaft speed and structure resonance. It is appropriate to identify ship by line spectra. In this study, a ship noise model has been used by combining line spectrum and continuous spectrum of a ship. Then, artificial neural network (ANN) has been used to calculate the contents of the line spectrum of a ship noise in order to identify a ship. Furthermore, in order to identify ship correctly, the received signal is restored to the source signal. Before signal restoration, the source needs to be located. Hence the signal restoration and source localization are the important issues for ship identifications. Finally, adaptive time reversal (ATR) is used to locate source and to restore signal. The results indicate that ATR is effective in source localization and source signal restoration. Therefore, it shows great advantages in identifying a ship

2:55

3pSA5. Extraction of independent non-Gaussian components from non-Gaussian background. Zbyněk Koldovský (Inst. of Information Technol. and Electronics, Tech. Univ. of Liberec, Studentska 2, Liberec 46117, Czechia, zbynek.koldovsky@tul.cz)

Independent component analysis (ICA) is a popular method for Blind Signal Separation applied to multichannel linear recordings. Convolutive mixtures of acoustic signals can be separated by applying ICA in the frequency domain to each sub-band separately or jointly as in Independent Vector Analysis (IVA). However, this way, the mixtures are separated into that many components as is the number of sensors. ICA and IVA are therefore not effective in applications where only one (or few) signals are of practical interest. We introduce a new parameterization of the ICA mixing model that is optimized for the extraction of one signal of interest (SOI). The approach is called Independent Component Extraction (ICE) and is closely related to methods for Blind Signal Extraction (BSE) such as One-unit FastICA. However, BSE methods assume that background signals (the other signals than SOI) are all Gaussian, which limits their accuracy. In this work, we show that ICE can be extended also for a non-Gaussian background so that the accuracy of algorithms is improved, although all components are not separated as in ICA. We also introduce an extension for the extraction of a vector component, so-called Independent Vector Extraction.

WEDNESDAY AFTERNOON, 9 MAY 2018 — NICOLLET A, 1:00 P.M. TO 3:20 P.M.

Session 3pSC

Speech Communication: Tools and Technology for Speech Research (Poster Session)

Mary M. Flaherty, Chair

Psychology, SUNY Buffalo, 392 Park Hall, North Campus, Buffalo, NY 14260

All posters will be on display and all authors will be at their posters from 1:00 p.m. to 3:20 p.m.

Contributed Papers

3pSC1. Fearless steps: Advancing speech and language processing for naturalistic audio streams from Earth to the Moon with Apollo. John H. L. Hansen, Abhijeet Sangwan, Lakshmish Kaushik, and Chengzhu Yu (Jonsson School of Eng. & Comput. Sci., CRSS: Ctr. for Robust Speech Systems; UTDallas, CRSS: Ctr. for Robust Speech Systems, The University of Texas at Dallas; 800 W Campbell Rd., Richardson, TX 75080-3021, john.hansen@utdallas.edu)

NASA's Apollo program represents one of the greatest achievements of mankind in the 20th century. CRSS-UTDallas has completed an effort to digitize and establish an Apollo audio corpus. The entire Apollo mission speech data consists of well over ~100,000 hours. The focus of this effort is to contribute to the development of Spoken Language Technology based algorithms to analyze and understand various aspects of conversational speech. Towards achieving this goal, a new 30 track analog audio decoder was designed using NASA Soundscriber. We have digitized 19,000 hours of data from Apollo 11,13,1 missions: named "Fearless Steps". An automated diarization and transcript generation solution was developed based on deep neural networks (DNN) automatic speech recognition (ASR) along with Apollo mission specific language models. Demonstration of speech technologies including speech activity detection (SAD), speaker identification (SID), and ASR are shown for segments of the corpus. We will release this corpus to the SLT community. The data provide an opportunity for challenging tasks in various SLT areas. We have also defined and proposed 5 tasks as a part of a community based SLT challenge. The five challenges are as follows: (1) automatic speech recognition, (2) speaker identification, (3) speech activity detection, (4) speaker diarization, and (5) keyword spotting and joint topic/sentiment detection. All data, transcripts, and guidelines for employing the fearless steps corpus will be made freely available to the community.

1:00

3pSC2. UTDallas-PLTL:Advancing multi-stream speech processing for interaction assessment in peer-led team learning. John H. L. Hansen, Harishchandra Dubey, Abhijeet Sangwan, Lakshmish Kaushik, and Vinay Kothapally (Jonsson School of Eng. & Comput. Sci., CRSS: Ctr. for Robust Speech Systems, UTDallas, The University of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080-3021, john.hansen@utdallas.edu)

Robust speech processing for single-stream audio data has achieved significant progress in the last decade. However, multi-stream speech processing poses new challenges not present in single-stream data. The peer-led team learning (PLTL) is a teaching paradigm popular among US universities for undergraduate education in STEM courses. In collaboration with UTDallas Student Success Center, we collected CRSS-PLTL and CRSS-PLTL-II corpora for assessment of speech communications in PLTL sessions. Both corpora consist of longitudinal recordings of five teams studying undergraduate Chemistry and Calculus courses consisting of 300 hours of speech data. The multi-stream audio data has unique challenges: (i) time-synchronization; (ii) multi-stream speech processing for speech activity detection, speaker diarization and linking, speech recognition, and (iii) behavioral informatics. We used a 1 kHz tone at the start and end of each session for time-synchronization of multi-stream audio. We leveraged auto-encoder neural network for combining MFCC features from multiple streams into compact bottleneck features. After diarization, each speaker segment is analyzed for behavioral metrics such as (i) dominance; (ii) curiosity in terms of question inflections; (iii) speech rate; (iv) cohesion; and (v) turn-duration and turn-taking patterns. Results are presented on individual and team based conversational interactions. This research suggests new emerging opportunities for wearable speech systems in education research.

3pSC3. Estimation of emotional arousal from speech with phase-based features. Igor Guoth, Sakhia Darjaa, Marian Trnka, Milan Rusko, Marian Ritomský (Inst. of Informatics, Slovak Acad. of Sci., Dubravska cesta 9, Bratislava 845 07, Slovakia, igor.guoth@savba.sk), and Roman Jarina (Univ. of Zilina, Zilina, Slovakia)

The most commonly adopted approaches in speech emotion recognition (SER) utilize magnitude spectrum and nonlinear Teager energy operator (TEO) based features while information about phase spectrum is often omitted. The information about phase has been frequently overlooked in approaches applied by speech processing researchers due to the signal processing difficulties. We present study of two phase-based features: The relative phase shift (RPS) based features and modified group delay features (MODGDF) that represents phase structure of speech in the task of emotional arousal recognition. The evaluation is performed on the CRISIS acted speech database which allows us to recognize five levels of emotional arousal from speech. To exploit these features, we employ concept of deep neural network. The efficiency of the approaches based on features mentioned earlier is compared to baseline platform using Mel frequency cepstral coefficients (MFCCs) and all pole group delay features (APGD). The combination of another phase-based types of features with our baseline platform led to the overall improvement of performance of the system for different levels of emotional arousal. These results confirm that combination of phase information and magnitude information leads to the overall improvement of performance of such system and also that combination of different types of features representing phase information brings additional increment of the performance.

3pSC4. The speakers in the room corpus. Aaron Lawson (SRI Int., 333 Ravenswood Ave., Menlo Park, CA 94025, aaron.lawson@sri.com), Karl Ni (Lab41, Menlo Park, CA), Colleen Richey, Zeb Armstrong, Martin Graciarena (SRI Int., Menlo Park, CA), Todd Stavish, Cory Stephenson, Jeff Hetherly, Paul Gamble, and Maria Barrios (Lab41, Menlo Park, CA)

The speakers in the room (SITR) corpus is a collaboration between Lab41 and SRI International, designed to be a freely available data set for speech and acoustics research in noisy room conditions. The main focus of the corpus is on distant microphone collection in a series of four rooms of different sizes and configurations. There are both foreground speech and background adversarial sounds, played through high-quality speakers in each room to create multiple, realistic acoustic environments. The foreground speech is played from a randomly rotating speaker to emulate head motion. Foreground speech consists of files from LibriVox audio collections and the background distractor sounds will consist of babble, music, HVAC, TV/radio, dogs, vehicles, and weather sounds drawn from the MUSAN collection. Each room has multiple sessions to exhaustively cover the background foreground combinations, and the audio is collected with twelve different microphones (omnidirectional lavalier, studio cardioid, and piezoelectric) placed strategically around the room. The resulting data set was designed to enable acoustic research on event detection, background detection, source separation, speech enhancement, source distance, sound localization, as well as speech research on speaker recognition, speech activity detection, speech recognition, and language recognition.

3pSC5. Creation and characterization of an emotional speech database. Peter M. Moriarty, Michelle Vigeant (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, moriarty@psu.edu), Rachel Wolf (Psych., Penn State Univ., University Park, PA), Rick Gilmore, and Pamela Cole (Psych., Penn State Univ., State College, PA)

Paralinguistic features of speech communicate emotion in the human voice. In addition to semantic content, speakers imbue their messages with prosodic features comprised of acoustic variations that listeners decode to extract meaning. Psychological science refers to these acoustic variations as *affective prosody*. Most studies of affective prosody obscure semantic content, although the resulting stimuli are less representative of naturally occurring emotional speech. The presented works details the creation of a naturalistic emotional speech database on which both acoustical analysis and a listening study were conducted. To this end, 55 adults were recorded speaking the same semantic content in happy, angry, and sad voices. Based on prior acoustic analyses of affective prosody, classic parameters were extracted including pitch, loudness, timing, as well as other low-level descriptors, and compared the acoustic features of each emotion with published evidence and theory. Preliminary results indicate that this naturalistic speech samples yielded acoustic features that are congruent with prior experimental stimuli of anger and happiness, but was less consistent with sadness. The results of the listening study indicated that listeners discriminated the intended emotions with 92% accuracy. The dataset therefore yielded a database of emotionally salient acoustical information for further analyses. [Work supported by NIH-R21-104547.]

3pSC6. Toward the automatic detection of manually labeled irregular pitch periods. Olivia Murton (Speech and Hearing BioSci. and Technol., Harvard Med. School, 1 Bowdoin St, #11, Boston, MA 02114, omurton@g.harvard.edu), Jeung-Yoon Choi (Massachusetts Inst. of Technol., Cambridge, MA), Daryush Mehta (Speech and Hearing BioSci. and Technol., Harvard Med. School, Boston, MA), and Stefanie Shattuck-Hufnagel (Massachusetts Inst. of Technol., Cambridge, MA)

Irregular pitch periods (IPPs) occur in a wide variety of speech contexts and can support automatic speech recognition systems by signaling word boundaries, phrase endings, and certain prosodic contours. IPPs can also provide information about emotional content, dialect, and speaker identity. The ability to automatically detect IPPs is particularly useful because accurately identifying IPPs by hand is time-consuming and requires expertise. In this project, we use an algorithm developed for creaky voice analysis by Kane *et al.* (2013) incorporating features from Ishi *et al.* (2008) to automatically identify IPPs in recordings of speech from the American English Map Task database. Short-term power, intra-frame periodicity, inter-pulse similarity, subharmonic energy, and glottal pulse peakiness measures are input into an artificial neural network to generate frame-by-frame creak probabilities. To determine a perceptually relevant threshold probability, the creak probabilities are compared to IPPs hand-labeled by experienced raters. Preliminary results yielded an area under the receiver operating characteristic curve of 0.82. Thresholds above 0.1 produced very high specificity, but even lower thresholds yielded fairly high sensitivity and specificity. These results indicate generally good agreement between hand-labeled IPPs and automatic detection, calling for future work investigating effects of linguistic and prosodic context.

3pSC7. Duration of connected speech needed to accurately estimate the articulatory-acoustic vowel space of a reading passage. Jason A. Whitfield, Anna Gravelin, Zoe Kriegel (Commun. Sci. and Disord., Bowling Green State Univ., 200 Health and Human Services Bldg., Bowling Green State University, Bowling Green, OH 43403, jawhitf@bgsu.edu), and Daryush Mehta (Dept. of Surgery–MGH, Harvard Med. School, Boston, MA)

Unlike other vowel space metrics, the articulatory-acoustic vowel space (AAVS) is calculated from the generalized variance of continuously sampled formant (F1-F2) traces. Given a sufficiently long speech sample, the AAVS should stabilize, though the duration required for the measure to converge remains unknown. The current investigation aimed to determine the amount of formant data needed for the AAVS to stabilize. Formant traces (20 ms frames, 10 ms intervals) were extracted using a Kalman-based autoregressive approach from readings of the Caterpillar passage produced by 16 speakers using habitual speech. The absolute percent difference (error) between cumulative AAVS estimates and the passage AAVS was calculated by iteratively adding F1-F2 pairs from successive voiced frames. Power functions were fit to the error plots using a least absolute residuals method to determine the frame at which the upper bound of the function fell within 5% of the passage AAVS. Across all speakers, the AAVS converged within 1065 voiced frames (i.e., 10.65 seconds of voiced speech). In terms of absolute speaking duration, all samples converged within 18 seconds (mean = 8.26; SD = 5.04). These data suggest that 15 to 20 seconds of connected speech is sufficient to provide a reasonable estimate of working vowel space using the AAVS.

3pSC8. Simulation of computerized screening for phonological errors in spoken utterances. Amitava Biswas, Anita Thames, and Steven Cloud (Speech and Hearing Sci., Univ. of Southern MS, 118 College Dr. #5092, USM-CHS-SHS, Hattiesburg, MS 39406-0001, Amitava.Biswas@usm.edu)

Fast and reliable diagnosis of phonological errors is often necessary. This presentation describes a simulation of computerized screening for phonological errors in spoken utterances. A set of common words will be included in a database. Every word in the database will be linked to several other words according to minimal pairs. For example, CAT will be linked to KATE, CUT, CAB, CASH, KITE, BAT, FAT, HAT, MAT, RAT, etc, whereas, RAT will be linked to CAT, RATE, WRITE, RASH, RAM, RAN, RACK, RAG, RAP, etc., and RAN will be linked to RAT, PAN, RAM, BAN, CAN, RUN, RAIN, etc. A set of hypothetical patterns of phonological errors in articulation will also be included in the database. The simulation will explore efficacy of navigating through the data base with the help of a custom software in predicting and correctly identifying the particular patterns of phonological errors.

3pSC9. Google speech recognition of an English paragraph produced by Korean college students in clear or casual speech styles. Byunggon Yang (English Education, Pusan National Univ., 30 Changjundong Keumjunggu, Pusan 609-735, South Korea, bgyang@pusan.ac.kr)

These days voice models of automatic speech recognition software are sophisticated enough to process the natural speech of people without any previous training. However, not much research has reported on the use of speech recognition tools in the field of pronunciation education. This paper examined Google speech recognition of a short English paragraph produced by Korean college students in clear and casual speech styles in order to diagnose and resolve students' pronunciation problems. Thirty three Korean college students participated in the recording of the English paragraph. The Google soundwriter was employed to collect data on the word recognition rates of the paragraph. Results showed that the total word recognition rate was 73% with a standard deviation of 9.1%. The word recognition rate of clear speech was around 77.3% while that of the casual speech amounted to 68.7%. The reasons for the low recognition rate of casual speech were attributed to both individual pronunciation errors and the software itself as shown in its fricative recognition. Various distributions of unrecognized words were observed depending on each participant and proficiency groups. From the results, the author concludes that the speech recognition software is useful to diagnose each individual or group's pronunciation problems.

3pSC10. Automated formant tracking using reassigned spectrograms. Sean A. Fulop (Linguist, California State Univ. Fresno, 5245 N Backer Ave., PB92, Fresno, CA 93740-8001, sfulop@csufresno.edu) and Christine H. Shadle (Haskins Labs., New Haven, CT)

The measurement and tracking of formant frequencies is usually accomplished with a linear prediction spectrum estimate, combined with a heuristic-based tracking algorithm. Despite the broad acceptance of this procedure over decades, there is a known bias towards the nearest harmonic, and formant estimates are less accurate for speech with higher fundamental frequency. Previous research has established the superior accuracy of formant measurement using the reassigned spectrogram; however, the measurements must be done by hand, severely limiting its usefulness. Here we present a technique that can locate formants automatically in a reassigned spectrogram. First, the reassigned spectrogram of the signal must be heavily "pruned" so that only a few points remain which highlight the detected frequency components, which are the presumed formants. Second, we apply a simple ridge-finding routine to the pruned spectrogram, to determine the formant tracks automatically. This process was tested on speech samples—/hVd/-words—by four speakers previously used in a comparison of 5 automatic algorithms and manually-measured reassigned spectrograms, and on breathy and falling tones in a speaker of Hmong. The results of this process are generally as good or better than the manual measurement when the optimal parameters for ridge-finding are used.

3pSC11. Effects of a mask rim leak on AC airflow measurement. Nicholas A. May and Ronald Scherer (Commun. Sci. & Disord., Bowling Green State Univ., 103 South Main St., Apt. 10, Bowling Green, OH 43402, nmay@bgsu.edu)

Studies in airflow during speech production typically use a pneumotachographic mask system to measure expired airflows. Accurate measures of airflow using the Glottal Enterprises MSIF-2 aerodynamic system require a complete seal of the mask rim to the face. Literature frequently cites mask rim leaks as causing flow measure inaccuracies, but quantitative studies of inaccuracies are needed. Prior work (May & Scherer, in press) provided a general empirical equation relating mask rim leak flow, the cross sectional area of the rim leak, and upstream airflow. The current empirical bench research extends this DC flow work to an AC flow situation as in the case of vibrato and/or tremor using 3 and 6 Hz, 2 leak areas, 50 and 200 cm^3/s peak-to-peak variations, and an upstream flow of 200 cm^3/s. Results: smaller leak area, higher AC frequency, and higher AC airflow extent resulted in the lowest mask rim leak airflow.

3pSC12. A method for quantifying vowel change directionality. Miranda R. Morris (Linguist, UC Davis, 1 Shields Ave., Davis, CA 95616, mirmor@ucdavis.edu)

Vowels are a common variable used in studies on phonetic accommodation and imitation. The standard practice for measuring changes in vowels provides an absolute value of the distance between a baseline vowel production and a target relative to the distance between shifted/imitated vowels and that same target. While distance alone is useful, a vector based approach would form a more complete picture as capturing distance alone can mask phenomena such as exaggeration. Vowels are a two-dimensional variable; thus, a directionality measure for changes in vowels relative to some target is crucial for fully capturing phenomena in imitation. Proposed is a trigonometric approach to quantifying the directionality of vowel changes. While this method was created to aid in studies on phonetic imitation, it is also potentially useful for any work involving vowel change with a known acoustic target.

3pSC13. Evaluating the LENA recording system for investigating speech input in a French-English bilingual context. Adriel John Orena (School of Commun. Sci. and Disord., McGill Univ., 2001 McGill College, 8th Fl., Montréal, QC H3A 1G1, Canada, adriel.orena@mail.mcgill.ca), Linda Polka (School of Commun. Sci. and Disord., McGill Univ., Montreal, QC, Canada), and Julia Srouji (Dept. of Psych., McGill Univ., Montréal, QC, Canada)

The Language ENvironment Analysis (LENA) recording system is rapidly gaining acceptance as a useful tool for assessing a child's speech input. The success of this tool is due to LENA algorithms which provide automated measures of the child's auditory environment, including the amount of words. The adult word count (AWC) measure has been validated in several monolingual language contexts, including English, French, Dutch, and Mandarin. Here, we evaluate LENA performance in counting words when two languages are present within the child's input stream. Twenty French-English bilingual families with a 10-month-old infant contributed full-day recordings. Nine 5-minute segments from each family was transcribed, for a total of 900 minutes of transcribed speech. We assessed the accuracy of LENA-generated AWC for French-English bilingual speech by comparing them to human-transcribed word counts. Linear mixed modelling reveals a positive linear relationship between these two measures. Critically, the correlations between these two measures were strong for both French and English, and these correlations were not significantly different from one another. These results confirm that the LENA algorithms are sufficiently accurate in counting words in two languages, and provide support for researchers and clinicians wishing to use the LENA recording system to assess bilingual speech.

WEDNESDAY AFTERNOON, 9 MAY 2018 GREENWAY H/I, 1:00 P.M. TO 3:00 P.M.

Session 3pSP

Signal Processing in Acoustics and Underwater Acoustics: Co-Prime Arrays and Other Sparse Arrays II

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

Kainam T. Wong, Cochair
Dept. of Electronic & Information Engineering, Hong Kong Polytechnic University, 11 Yuk Choi Road, Hung Hom KLN, Hong Kong

Invited Papers

1:00

3pSP1. Biomedical applications of sparse hemispherical ultrasound arrays. Ryan M. Jones, Meaghan A. O'Reilly (Medical Biophys., Univ. of Toronto, 2075 Bayview Ave., Focused Ultrasound Lab (C713), Toronto, ON M4N 3M5, Canada, rmjones@sri.utoronto.ca), Lulu Deng (Physical Sci. Platform, Sunnybrook Res. Inst., Toronto, ON, Canada), and Kullervo Hynynen (Medical Biophys., Univ. of Toronto, Toronto, ON, Canada)

Sparse arrays have been employed in biomedical ultrasound for diagnostic and therapeutic purposes to reduce the technical complexities associated with dense arrays while maintaining adequate performance characteristics. Sparse arrays are particularly attractive when large apertures are desired, which is often the case for therapy to achieve high acoustic gains and tight focal volumes. Large aperture arrays provide similar benefits on receive during multichannel beamforming of passively detected acoustic emissions, which is a promising approach for guiding cavitation-mediated ultrasound therapies. Our group has designed and fabricated sparse hemispherical phased arrays for ultrasound brain imaging and therapy. Ultrasound field simulations were carried out to optimize the spatial distribution of array elements. We have employed these devices for skull computed tomography-array registration, 3D spatial mapping of microbubble activity *in vivo* through *ex-vivo* human skullcaps via noninvasive aberration correction methods, and have harnessed this spatiotemporal cavitation information to calibrate exposure levels for safe, volumetric ultrasound-induced blood-brain barrier opening. In conjunction with techniques borrowed from optical microscopy, we have demonstrated trans-skull 3D microbubble imaging beyond the diffraction limit, which may have application in ultrasound-based cerebral angiography. This talk will review our progress to date in the development and biomedical application of sparse hemispherical ultrasound arrays.

1:20

3pSP2. Passive acoustic mapping and B-mode ultrasound imaging utilizing compressed sensing for real-time monitoring of cavitation-enhanced drug delivery. Calum Crake, Seán Finn, Laurent Marsac (OxSonics Ltd., OxSonics Ltd., The Magdalen Ctr., Robert Robinson Ave., Oxford OX4 4GA, United Kingdom, calum.crake@gmail.com), Michael Gray, Robert Carlisle, Constantin Coussios (Univ. of Oxford, Oxford, United Kingdom), and Christian Coviello (OxSonics Ltd., Oxford, United Kingdom)

Ultrasound imaging presents a high-speed, low-cost approach for monitoring of focused ultrasound (FUS) therapy, and includes both conventional (B-mode) sonography and passive acoustic mapping (PAM) of acoustic emissions (Gyöngy *et al.*, 2010, Salgaonkar *et al.*, 2009). Incorporation of novel algorithms and other signal processing techniques have improved the resolution and processing speed of PAM. However, while hardware developments such as increasing channel counts provide unprecedented data capture ability, real-time processing of the growing data stream presents an evolving challenge. Previous work has employed sparse array processing techniques for PAM including matching and basis pursuit (Gyöngy & Coviello, 2011) and co-array processing (Coviello et al, 2012). Here we propose to extend PAM utilizing compressed sensing (CS). Acoustic emissions from FUS may be sparse in several domains, e.g. due to limited regions of space and time in which cavitation is likely from a focused transducer, and correlation between sensors. By exploiting this sparsity, CS allows recovery of signals sampled well below the Nyquist limit. CS-PAM thus facilitates PAM with improved spatial resolution compared to conventional methods, from fewer measurements, allowing improved image quality and reduced computational load. The technique was demonstrated for monitoring of FUS-mediated drug delivery in a murine tumor model.

1:40

3pSP3. Separation of closely-spaced acoustics sources in an under-determined system with convex optimization. Tongyang Shi and J. S. Bolton (Mechanical Eng., Purdue Univ., Ray W. Herrick Labs., 177 S Russell St., West Lafayette, IN 47907, shi247@purdue.edu)

The ability to identify acoustical source locations accurately is critical when performing sound field reconstructions. In previous work, a monopole-based, iterative equivalent source method, wideband acoustical holography, has proven able to provide accurate noise source location in complex machines even when the number of measurements was far smaller than the number of parameters that needed to be determined in the model (see: Tongyang Shi, Yangfan Liu and J. Stuart Bolton, "Diesel engine noise source visualization with wideband acoustical holography," SAE paper 2017-01-1874). However, at some stage, when acoustical sources are too closely spaced, the current algorithm has difficulty in separating them. This is true, especially at low frequencies, in which case current holography methods tend to visualize two closely-spaced acoustics sources as a single large source, thus losing the accuracy of the source location. In the present work, the equivalent sources were still modeled as an array of monopoles, but the source strength parameter estimation and regularization problem was formulated as a convex function and turned into an iterative convex optimization problem that can be solved by an open source code. It will be demonstrated that the new procedure is capable of reconstructing very closely-spaced acoustics sources.

2:00

3pSP4. Highly directional pressure sensing using phase gradients. Joseph S. Lawrence, Kent L. Gee, Tracianne B. Neilsen, and Scott D. Sommerfeldt (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, joseph-lawrence@hotmail.com)

The recently developed Phase and Amplitude Gradient Estimator (PAGE) method [Thomas *et al.*, J. Acoust. Soc. Am. **137**, 3366–3376 (2015)] estimates active intensity with a much greater bandwidth of accuracy than traditional intensity estimates. In this formulation, the intensity direction comes from the phase gradient, which when estimated from unwrapped transfer function phases can be accurate above the spatial Nyquist frequency. The phase gradient term can also be used to express pressure as a sum of directional components, which is derived by using the continuity equation with Euler's equation. These directional pressure components have a cosine-squared directivity, which can be increased to higher orders by raising the power of the phase gradient. This method allows for the creation of highly directional pressure sensors in one or multiple dimensions using a limited number of microphones. [Work supported by NSF.]

2:20

3pSP5. Azimuth-elevation direction finding, using three uni-axial velocity sensors as an arbitrarily sparse linear array. Yang Song, Kainam T. Wong, Salman Khan, and Mohammad Asaduzzaman Khan (Dept. of Electron. and Information Eng., The Hong Kong Polytechnic Univ., CD515, Hung Hom, Kowloon, Hong Kong., Hung Hom KLN, Hong Kong, salman.khan@connect.polyu.hk)

A tri-axial velocity sensor measures the acoustic particle-velocity vector, by all three of its Cartesian components. This particle-velocity vector equals the spatial gradient of the acoustic pressure field. Song & Wong [1] has advanced an algorithm to estimate an incident source's azimuth-elevation (θ, Φ) direction-of-arrival using three uni-axial velocity sensors that are orthogonally oriented among themselves and placed arbitrarily in three-dimensional space. This work will focus on a particular array geometry whereby the three uni-axial velocity sensors are placed on a straight line in three-dimensional space but the inter-element spacings may be entirely arbitrary. This work will show how to estimate a far-field emitter's azimuth-elevation direction-of-arrival, despite the three elements' spatial separation, and despite the separation's arbitrary length and possible sparseness. Furthermore, the Cramer-Rao Bounds (CRB) will be presented analytically for the case of the linear array being aligned along the x-axis. This work differs from [2], which requires an additional pressure-sensor. [1] Y. Song & K. T. Wong, "Acoustic direction finding using a spatially spread tri-axial velocity sensor," IEEE Trans. Aerosp. Electron. Syst. **51**(2), 834–842 (2015). [2] Y. Song & K. T. Wong, "Azimuth-elevation direction finding, using one four-component acoustic vector-sensor spread spatially along a straight line," Proc. of Meetings on Acoustics—Meeting of the ASA, Pittsburgh, U.S.A., May 18–22, 2015.

2:40

3pSP6. Identification of fin whale 20-Hz pulses and other vocalizations from passive monitoring with a coherent hydrophone array. Wei Huang (Elec. and Comput. Eng., Northeastern Univ., 500 Broadway, Apt 3157, Malden, MA 02148, huang.wei1@husky.neu.edu), Heriberto A. Garcia (Elec. and Comput. Eng., Northeastern Univ., Arlington, MA), and Purnima Ratilal (Elec. and Comput. Eng., Northeastern Univ., Boston, MA)

A large variety of sound sources in the ocean, including biological, geophysical and man-made activities can be simultaneously monitored over instantaneous continental-shelf scale regions via the passive ocean acoustic waveguide remote sensing (POAWRS) technique by employing a large-aperture densely-sampled coherent hydrophone array. Millions of acoustic signals received on the POAWRS system per day can make it challenging to identify individual sound sources. An automated classification system is necessary to enable sound sources to be recognized. Here each detected acoustic signal is represented by an amplitude weighted pitch track which describes its fundamental frequency-time and amplitude variation. Multiple features are extracted from the pitch track including mean, minimum and maximum frequencies, bandwidth, signal duration, frequency-time slope and curvature, as well as several amplitude weighted features. A large training data set of fin whale 20-Hz pulses and other vocalizations are gathered after manual inspection and labeling. Next, multiple classifiers including logistic regression and decision tree are built and tested for identifying the fin whale 20-Hz pulses and other vocalizations from the enormous amounts of acoustic signals detected per day. The performance of the classifiers are evaluated and compared

WEDNESDAY AFTERNOON, 9 MAY 2018 GREENWAY B, 1:00 P.M. TO 2:05 P.M.

Session 3pUW

Underwater Acoustics: Target Scattering in Underwater Acoustics: Imaging, Spectral Domain, and Other Representations II

Daniel Plotnick, Cochair
Applied Physics Lab., Univ. of Washington, Seattle, WA 98105

Timothy Marston, Cochair
APL-UW, 1013 NE 40th Street, Seattle, WA 98105

Chair's Introduction—1:00

Contributed Papers

1:05

3pUW1. Shallow water multichannel sub-bottom profiling using Kirchhoff imaging. João P. Ristow, Marina M. Bousfield, Gregório G. Azevedo (Mech. Eng., Federal Univ. of Santa Catarina, Rua Lauro Linhares, 657, Apto. 203B, Florianópolis, Santa Catarina 88036-001, Brazil, jpristow@gmail.com), Julio A. Cordioli (Mech. Eng., Federal Univ. of Santa Catarina, Florianópolis, SC, Brazil), and Stephan Paul (Mech. Eng., Federal Univ. of Santa Catarina, Joinville, Brazil)

Single-channel sub-bottom profilers are commonly used for shallow water seismic investigation. This work reports the design and construction of a multiple channel profiler that allows more effective seismic imaging techniques. The system is composed of a controlled acoustic source and a 1.6 m long array with eight evenly spaced receiver channels, and operates using CHIRP signals with frequency ranging from 800 Hz to 8 kHz. A survey was carried out to obtain seismic data in the region of Lagoa da Conceição, Santa Catarina, Brazil. The acquired data was processed using a seismic imaging technique based on the pre-stack Kirchhoff time migration. Both the multichannel seismic data acquisition system and the applied imaging technique showed to be robust for acquiring subsurface information. The use of the pre-stack Kirchhoff time migration resulted in a seismic image with reflection events allocated in the real position and without bowtie effects since it repositions the diffracted energy back to the spreaders that originated it. This procedure results in a more reliable subsurface image of the studied environment, when compared with single-channel sub-bottom profilers.

1:20

3pUW2. Effect of mesoscale eddies on deep-water sound propagation. Yao Xiao (Inst. of Acoust., Chinese Acad. of Sci., 801 Ferst Dr., George W. Woodruff School of Mech. Eng., Atlanta, GA 30332, xyao62@gmail.com), Zhenglin Li (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China), and Karim G. Sabra (Mech. Eng., Georgia Tech, Atlanta, GA)

The mesoscale eddies are the most common mesoscale phenomena in the ocean. The cold and warm water carried by the mesoscale eddies can significantly change the structure of the sound speed field, which is the key factor affecting long range deep-water sound propagation. In this presentation, a deep-sea analytical eddy model [Henrick *et al.*, JASA **62**, 860–870(1977)] is used to determine the sound speed distributions produced by cold eddy in the southwest of the South China Sea. The statistical characteristics of the influence of eddies on the acoustic propagation are investigated with a range-dependent parabolic equation (PE) model to asses the effects of the relative position of sound sources and eddies, the frequency sensitivity, the acoustic energy deviation of the convergence zone and the shadow zone for long-range (600 km) low frequency (50 Hz) sound propagation in

this ocean model. Specific features of the induced sound propagation variability such as the travel time deviation and the standard deviation of ensemble intensity Will be quantified. Finally, the refraction due to mesoscale eddies on large scale low-frequency propagation is discussed using the full three-dimensional parabolic (FOR3D) and explaining by the full three-dimensional ray theory (Bellhop3D).

1:35

3pUW3. Scattering of low-frequency sound by infinite fluid and solid cylinders. Alexander B. Baynes and Oleg A. Godin (Dept. of Phys., Naval Postgrad. School, Spanagel Hall, Rm. 215, Monterey, CA 93943-5216, abbaynes1@nps.edu)

Wave scattering by obstacles is typically studied assuming plane wave incidence. However, full Green's functions are necessary for problems where the separation of the scatterer from sources, interfaces, or other scatterers is comparable to the dimensions of the scatterer itself. In this paper, the two-dimensional problem of scattering of monochromatic cylindrical waves due to infinite cylinder embedded in a homogeneous fluid is considered. Fluid and solid cylinders are studied, and soft and hard cylinders are revisited. The exact solutions for the Green's functions are expressed as an infinite series of cylindrical functions with complex amplitudes determined by the acoustic boundary conditions at the surface of the cylinder. Here, we derive closed-form asymptotics for the scattered field in the regime when the scatterer dimensions are small compared to wavelength, i.e., Rayleigh scattering. The scattered wave approximation is valid for arbitrary source and observation points outside the scatterer and is expressed as a simple sum of fields due to three image sources. Image source solutions were anticipated due to classically studied electrostatic analog problems involving dielectric cylinders. Image source representation allows physical insight into the scattering physics and suggests simple yet accurate analytic solutions to interface and waveguide scattering problems.

1:50

3pUW4. Scattering of low-frequency sound by shallow underwater targets. Alexander B. Baynes and Oleg A. Godin (Dept. of Phys., Naval Postgrad. School, Spanagel Hall, Rm. 215, Monterey, CA 93943-5216, abbaynes1@nps.edu)

Rayleigh scattering of sound by a target can be described as a wave radiated by virtual point sources inside the target. When a target is located close to the ocean surface or another reflecting boundary, reflections of the incident and single-scattered waves from the boundary lead to multiple scattering from the target, with the target being insonified by nearby virtual sources. At low frequencies and for shallow targets, the distance from a virtual source to the target is not necessarily large compared to acoustic wavelength or the target's dimensions. This paper takes advantage of the virtual source concept and recently derived explicit analytic representations of 2-D and 3-D acoustic Green's functions in unbounded fluids with inclusions of a circular cross-section, to develop a simple and numerically efficient model of multiple scattering. Scattering from soft, hard, fluid, and solid objects is considered. The model is used to study the acoustic field in the vicinity of shallow targets, calculate normal-mode compositions of the multiple-scattered field, examine conditions of applicability of the single-scattering approximation, and investigate implications of multiple scattering for target detection and classification.

Plenary Session and Awards Ceremony

Marcia J. Isakson,
President, Acoustical Society of America

Annual Membership Meeting

Presentation of Certificates to New Fellows

Deniz Baskent – For contributions to our understanding of acoustic and electric auditory and speech perception

Monita Chatterjee – For contributions to cochlear implant psychophysics and speech perception

Patricia Davies – For contributions to the fields of sound quality, aircraft noise, and the dynamic properties of foams

Jin-Yong Jeon – For contributions to research and practical design in soundscape and architectural acoustics

Bruce C. Olson – For contributions to design and modeling of electro-acoustic systems for architectural spaces

Lauri Savioja – For contributions to room-acoustics modeling and auralization

Christopher J. Struck – For contributions to acoustical standards development and instrumentation design

Introduction of Award Recipients and Presentation of Awards

William and Christine Hartmann Prize in Auditory Neuroscience to Shihab Shamma

Medwin Prize in Acoustical Oceanography to Yin-Tsong Lin

R. Bruce Lindsay Award to Yun Jing

von Békésy Medal to David Kemp

Helmholtz-Rayleigh Interdisciplinary Silver Medal to Kenneth S. Suslick

Gold Medal to William A. Yost

Vice President’s Gavel to Michael J. Buckingham

President’s Tuning Fork to Marcia J. Isakson

Session 3eED

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

Peggy B. Nelson, Cochair
Univ. of Minnesota, Minneapolis, MN 55455

Keeta Jones, Cochair
Acoustical Society of America, 1305 Walt Whitman Rd., Suite 300, Melville, NY 11787

This workshop for Minneapolis area Girl Scouts 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.

WEDNESDAY EVENING, 9 MAY 2018 7:30 P.M. TO 9:00 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings. All meetings will begin at 7:30 p.m., except for Engineering Acoustics which will hold its meeting starting at 4:30 p.m.

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

Committees meeting on Wednesday are as follows:

Biomedical Acoustics	Greenway F/G
Signal Processing in Acoustics	Greenway H/I

ACOUSTICAL SOCIETY OF AMERICA

R. BRUCE LINDSAY AWARD

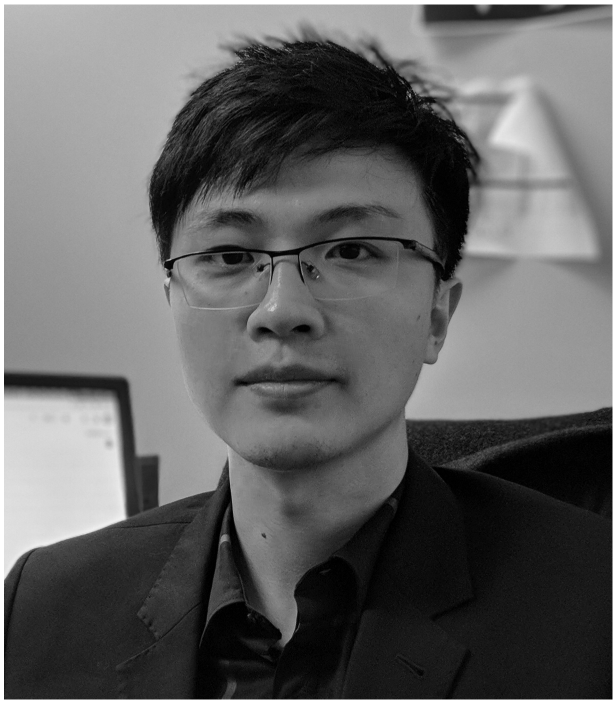
Yun Jing

2018

The R. Bruce Lindsay Award (formerly the Biennial Award) is presented in the Spring to a member of the Society who is under 35 years of age on 1 January of the year of the Award and who, during a period of two or more years immediately preceding the award, has been active in the affairs of the Society and has contributed substantially, through published papers, to the advancement of theoretical or applied acoustics, or both. The award was presented biennially until 1986. It is now an annual award.

PREVIOUS RECIPIENTS

Richard H. Bolt	1942
Leo L. Beranek	1944
Vincent Salmon	1946
Isadore Rudnick	1948
J. C. R. Licklider	1950
Osman K. Mawardi	1952
Uno Ingard	1954
Ernest Yeager	1956
Ira J. Hirsh	1956
Bruce P. Bogert	1958
Ira Dyer	1960
Alan Powell	1962
Tony F. W. Embleton	1964
David M. Green	1966
Emmanuel P. Papadakis	1968
Logan E. Hargrove	1970
Robert D. Finch	1972
Lawrence R. Rabiner	1974
Robert E. Apfel	1976
Henry E. Bass	1978
Peter H. Rogers	1980
Ralph N. Baer	1982
Peter N. Mikhalevsky	1984
William E. Cooper	1986
Ilene J. Busch-Vishniac	1987
Gilles A. Daigle	1988
Mark F. Hamilton	1989
Thomas J. Hofler	1990
Yves H. Berthelot	1991
Joseph M. Cuschieri	1991
Anthony A. Atchley	1992
Michael D. Collins	1993
Robert P. Carlyon	1994
Beverly A. Wright	1995
Victor W. Sparrow	1996
D. Keith Wilson	1997
Robert L. Clark	1998
Paul E. Barbone	1999
Robin O. Cleveland	2000
Andrew J. Oxenham	2001
James J. Finneran	2002
Thomas J. Royston	2002
Dani Byrd	2003
Michael R. Bailey	2004
Lily M. Wang	2005
Purnima Ratilal	2006
Dorian S. Houser	2007
Tyrone M. Porter	2008
Kelly J. Benoit-Bird	2009
Kent L. Gee	2010
Karim G. Sabra	2011
Constantin-C. Coussios	2012
Eleanor P. J. Stride	2013
Matthew J. Goupell	2014
Matthew W. Urban	2015
Megan S. Ballard	2016
Bradley E. Treeby	2017



CITATION FOR YUN JING

. . . for contributions to acoustic metamaterials and numerical modeling of wave propagation in rooms and complex media

MINNEAPOLIS, MINNESOTA • 9 MAY 2018

Yun Jing was born into an intellectual family in Nanjing, China. His maternal grandfather was a renowned Chinese scholar of linguistics, his mother is a novelist, and his father is a professor of Chinese literature. Unlike his parents and grandfather, Yun developed an interest in engineering rather than in literature. Yun chose acoustics as his undergraduate major and graduated in 2006. He earned M.S. and Ph.D. degrees in 2007 and 2009, from the Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute (RPI). He is currently a tenured Associate Professor in the Department of Mechanical Engineering, North Carolina State University (NCSU). Before joining the faculty at the NCSU, Yun also spent two years from 2009 to 2011 as a Research Fellow at Brigham and Women's Hospital at Harvard Medical School.

A few weeks after arriving at RPI in the fall of 2006, Yun asked his advisor for possible directions for his graduate research. When he was given a paper on experimental investigations using 3-D p-u sensors that was just published in the *Journal of the Acoustical Society of America* (JASA), Yun responded with moderate enthusiasm and asked for other options. Two days after he was given a more mathematical and theoretical JASA publication on diffusion theory in room acoustics, he wrote to his advisor that, "not only do I understand the underlying theory, but I also implemented the work in this paper and came up with exactly the same results as published." Two months later he demonstrated in a draft JASA letter that the current application of the diffusion equation in room acoustics can produce improved results using an Eyring boundary condition [Jing and Xiang, JASA **121**, 3284–3287, 2007]. He soon incorporated more rigorous boundary conditions into diffusion equation models, which led to his second peer-reviewed journal publication [Jing and Xiang, JASA **123**, 145–153, 2008], 10 months after his arrival at RPI. His work on the boundary conditions allows rigorous application of diffusion theory for efficient room acoustic predictions of sound energy flow and decay in enclosed spaces. Yun's accomplishments resulted in the successful completion of his M.S. degree in Architectural Acoustics, with two JASA publications in less than one year.

The transport theory has drawn attention in the urban noise community in the context of modeling street canyons since about 2004. However, no solution of the transport equation was available then that was useful for this application. Within the scope of his doctoral research, Yun developed a collaboration with Prof. Ed Larsen of the Department of Nuclear Engineering and Radiological Sciences at the University of Michigan. As a result, his thesis contains the methodology for solving the transport equation to numerically model room acoustics energy propagation, along with experimental validation of the theory implemented for long-space configurations. Thus, Yun produced pioneering work on transport equation-based room acoustic modeling, successfully finishing his Ph.D. dissertation at RPI in August, 2009, in less than two years. He and his advisors published seven JASA papers along this line of research.

Yun subsequently went further to pursue postdoctoral research at the Brigham and Women's Hospital of Harvard Medical School. He worked under Dr. Greg Clement, studying extensions of the angular spectrum approach for modeling of nonlinear ultrasonic wave propagation. This was a major shift in research direction from anything Yun had done in architectural acoustics. Nonetheless, Yun was never one to shy away from taking on new challenges such as understanding nonlinear acoustics, with gratitude to Hamilton and Blackstock's book. Yun and Greg discovered that the Westervelt equation in the wave-number-frequency-domain, can be solved using a one-dimensional Green's function. This allowed them to propose and investigate a modified angular spectrum approach. Following his initial ultrasonic work, Yun became intrigued by numerical modeling of ultrasound wave propagation, leading him to publish several other journal papers on this topic. For example, he developed another algorithm for solving the Westervelt equation based on the wavenumber-space method. It has the advantage that medium heterogeneities can be taken into account to model more realistic biological tissue. Subsequently, Yun used this method for finding the phase correction of transcranial beam focusing, which he continues to work on because of its usefulness in medical ultrasound.

At NCSU Yun actively pursues work in architectural acoustics and numerical modeling of ultrasound wave propagation. In addition, he has branched out into two different areas, acoustic metamaterials and therapeutic ultrasound. Yun realized the excitement and rapid growth in the field of metamaterials, which provided him with many opportunities to explore. His group recently published 13 journal articles on this topic, including four in two of the most prestigious physics journals, *Physical Review Letters* (PRL) and *Physical Review X*. One of his PRL papers was selected as an "Editor's Suggestion" [*PRL* **115**, 254301, 2015]. In this paper, Yun's group proposed a simple structure consisting of paper and aluminum that exhibits a negative effective density in one direction and positive effective density in another, giving rise to a peculiar hyperbolic dispersion. Such a hyperbolic metamaterial can be used for applications such as energy focusing, tunneling, and subwavelength imaging. This is the first time that a broad-band, truly hyperbolic acoustic metamaterial was proposed and experimentally validated.

Yun plays an active role in the affairs of the Society. In addition to attendance with paper presentations at meetings of the Acoustical Society of America (ASA) since 2006, he has been invited many times to deliver invited papers at ASA meetings and other conferences, he has recently organized and chaired three special sessions and chaired a number of other sessions during the ASA meetings.

Dr. Yun Jing is an exceptionally creative young scientist who has demonstrated a track record of high creativity and self-direction. Yun has made an important impact on the state of knowledge regarding numerical modeling of wave propagation in complex media and acoustic metamaterials. We are delighted to see the Acoustical Society of America recognizing and honoring Dr. Yun Jing with its prestigious R. Bruce Lindsay Award.

NING XIANG
WILLIAM SIEGMANN

ACOUSTICAL SOCIETY OF AMERICA

VON BÉKÉSY MEDAL

David Kemp

2018

The von Békésy Medal is presented to individuals, irrespective of nationality, age, or society affiliation, who have made outstanding contributions to the area of psychological or physiological acoustics as evidenced by publication of research results in professional journals or by other accomplishments in the fields.

PREVIOUS RECIPIENTS

Jozef J. Zwislocki	1985
Peter Dallos	1995
Murray B. Sachs	1998
William S. Rhode	2010
M. Charles Liberman	2012

CITATION FOR DAVID T. KEMP

. . . for the discovery of otoacoustic emissions and contributions to cochlear biophysics and the detection of hearing loss

MINNEAPOLIS, MINNESOTA • 9 MAY 2018

David Kemp trained as a geophysicist, earning his Ph.D. in physics from the University of London in 1970. His early research focused on the prominent series of peaks in the extreme low-frequency (ELF) region of the Earth's electromagnetic spectrum known as "Schumann resonances." Schumann resonances are global electromagnetic standing waves, excited by lightning discharges, that form in the "cavity" bounded by the Earth's surface and the ionosphere. With this background, David was well poised to grasp the significance of the curious, quasi-periodic pattern of spectral peaks and valleys ("microstructure") evident in the hearing threshold curve. David immediately understood this microstructure as the tell-tale signature of standing waves inside the "cavity" of the cochlea. These standing waves, he realized, should be detectable outside the cavity, in the external ear canal, if only one could listen hard enough. And thus, by listening to the ear, David's soon-to-be-illustrious career in auditory neuroscience was born.

By having the genius to do the simple and obvious—placing a miniature low-noise microphone in the human ear canal and carefully recording and interpreting the ear's response to sound—David Kemp sparked nothing short of a revolution in our understanding of the physical and physiological basis of hearing. David's discovery in the mid-1970s that the ear makes sound while listening to sound completely overturned the reigning dogma of the day that the cochlea was nothing but a linear, passive mechanical transducer.

Early on, many refused to believe it, regarding the "Kemp effect" as some sort of insidious artifact. But these critics overlooked David's brilliant series of well-controlled experiments establishing beyond doubt that the sounds he was recording originated within the inner ear. Others, although willing to credit the result, failed to appreciate the extraordinary theoretical and practical importance of these sounds, which are now known as otoacoustic emissions. For this reason, David experienced difficulty getting his results published. The manuscript announcing the discovery was rejected by *Nature* on the grounds that otoacoustic emissions would surely prove of little interest outside the community of clinicians concerned with the diagnosis of hearing impairment. Thus, thanks to the obtuseness of *Nature*'s editors, the Acoustical Society of America (ASA) received the honor of publishing David's seminal results. David's initial report is now the fourth most-cited paper in the history of the *Journal of the Acoustical Society of America* (and, according to the Science Citation Index, is the only one in the top nine concerned with auditory physiology).

David's discovery and his subsequent studies probing the physical and physiological mechanisms confirmed controversial hints of cochlear mechanical nonlinearity and revived a largely forgotten suggestion—by astrophysicist Thomas Gold in 1948—that the cochlea employs "regenerative action" powered by electrochemical potentials to enhance cochlear frequency selectivity by counteracting the viscous damping. Thus, David's work launched a flurry of experimental and theoretical activity, and within just a few short years the paradigm had shifted completely—the inner ear was now no mere passive transducer, but an active nonlinear amplifier controlled via neural feedback from the brain. Researchers soon found that the cochlea contains an array of cellular force generators (outer hair cells) that act in concert to amplify quiet sounds, thereby boosting the sensitivity and dynamic range of hearing.

Today, the quest to understand the mechanical, cellular, and molecular basis of the "cochlear amplifier" and its feedback control mechanisms dominates the study of the peripheral auditory system, and understanding the role of nonlinear amplification and compression in the analysis and perception of sound is a central theme in psychophysics. In the years since Kemp's discovery, the revolution has spread throughout the animal kingdom. Otoacoustic emissions (OAEs) almost identical to those first discovered by Kemp have been reported in animals as diverse as birds, lizards, frogs, and insects. The biological amplification of sound appears to be a universal theme in sensory physiology.

All of this is the direct result of David's singular contribution to basic science. But David had the further insight that OAEs had enormous translational potential. He realized that the emitted sounds provide information about the health of the inner ear—because changes in emission magnitudes reflect changes in hearing sensitivity, OAEs can be used as an objective test for the presence of hearing impairment. David showed that OAEs could be recorded using a small probe—containing both an earphone to provide the stimulus sound and a microphone to detect the emission—inserted into the ear canal. Recording OAEs was therefore a simple, noninvasive procedure capable of detecting sound-evoked biological activity in the inner ear.

After validating this procedure as an objective test of auditory function—and having failed to obtain the necessary support, either from the hearing test industry or from government bodies—David established his own company, Otodynamics Ltd, to develop, manufacture, and refine the test equipment and its software. With the unwavering support of his wife, Gillian, David built the company while continuing his full-time academic commitments as a lecturer and then Professor of Auditory Biophysics at University College, London (UCL). The resulting "ILO88" instrument set the standard for what has become a vital piece of every audiologist's armamentarium. Today, OAE recording is an indispensable component of the battery of audiological tests used in hospitals and clinics around the world; literally thousands of ILO88s and its successors are testing hearing every day, in both advanced and developing countries. OAE recording has also become a routine procedure in basic science laboratories investigating the mechanisms underlying normal hearing and its pathologies.

As noninvasive tests that require no input from the patient, OAE measurements are particularly valuable for assessing the hearing of babies. David recognized this potential and saw the value of early hearing screening so that intervention for hearing impairment might be implemented early in life. He was at the forefront of the drive for screening all neonates for hearing impairment. Newborn screening was first implemented in the USA, later in Europe and the UK. The early identification of hearing impairment is, of course, of enormous significance for the social and educational development of affected children.

David has used resources from his company to help build a premier auditory research facility, the Centre for Auditory Research at the UCL Ear Institute. This facility now houses many of the world's top researchers devoted to the study of hearing in all its aspects, including auditory biophysics, psychophysics, neuro- and molecular biology, and genetics. Academically, David has led the development and teaching of undergraduate degree courses in Audiology and postgraduate degree programs in Audiological Science and Audiological Medicine, as well as training and mentoring the next generation of researchers in the field though his supervision of Ph.D. students and postdoctoral fellows. At hearing conferences, David is often sighted quietly haunting the poster aisles, patiently probing and encouraging those young researchers now following in his footsteps.

Over the years, David has received many honors and awards recognizing his outstanding contributions to basic and applied hearing science. These include the Association for Research in Otolaryngology's Award of Merit, the American Speech-Language-Hearing Association's Distinguished Service Award, and his election as a Fellow of the Royal Society. David is a supreme example of how science, technology, and innovation can combine for the betterment of society. He has played a leading role in developing the measurement and analysis of OAEs into a ubiquitous and invaluable tool for improving human health. We are very pleased to congratulate David Kemp for being awarded the Acoustical Society of America's von Békésy Medal.

CHRISTOPHER A. SHERA
ANDREW FORGE
BRENDA L. LONSBURY-MARTIN

ACOUSTICAL SOCIETY OF AMERICA
HELMHOLTZ-RAYLEIGH INTERDISCIPLINARY SILVER MEDAL
in

Physical Acoustics and Biomedical Acoustics

Kenneth S. Suslick

2018

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.

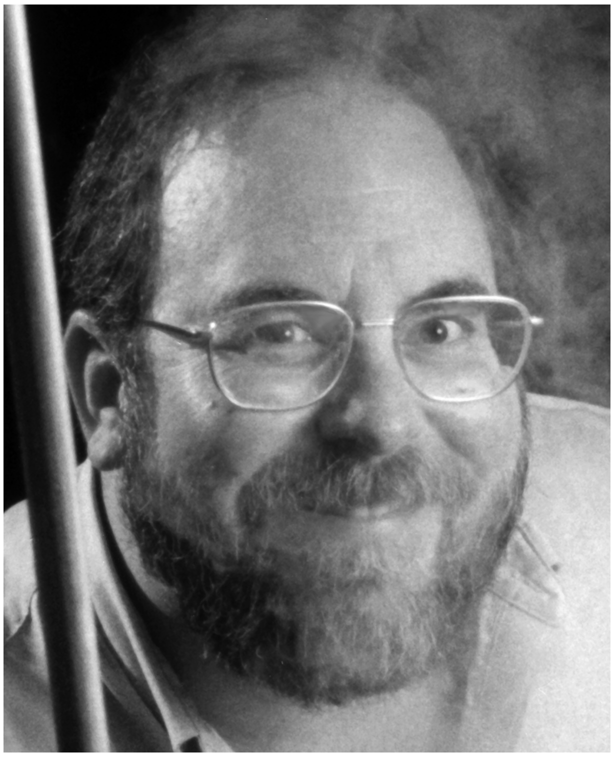
PREVIOUS RECIPIENTS

Helmholtz-Rayleigh Interdisciplinary Silver Medal

Gerhard M. Sessler	1997	Edwin L. Carstensen	2007
David E. Weston	1998	James V. Candy	2008
Jens P. Blauert	1999	Ronald A. Roy	2010
Lawrence A. Crum	2000	James E. Barger	2011
William M. Hartmann	2001	Timothy J. Leighton	2013
Arthur B. Baggeroer	2002	Mark F. Hamilton	2014
David Lubman	2004	Henry Cox	2015
Gilles A. Daigle	2005	Armen Sarvazyan	2016
Mathias Fink	2006	Blake S. Wilson	2017

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 KENNETH S. SUSLICK

. . . for contributions to the acoustics of sonochemistry

MINNEAPOLIS, MINNESOTA • 9 MAY 2018

Ken Suslick received his B.S. from the California Institute of Technology in 1974, his Ph.D. from Stanford University in 1978, and moved immediately to the University of Illinois where he is now Marvin T. Schmidt Research Professor of Chemistry. A prolific scientist, he has published nearly 400 papers, edited four books, and holds 48 patents and patent applications. As a modern measure of scientific productivity, he is listed under Google Scholar as having over 46,000 citations and an h-index of 109—which means 109 of his papers have been cited at least 109 times.

Among his numerous other research recognitions, He is the winner of many awards, among them: Fellow, National Academy of Inventors; Centenary Prize, Royal Society of Chemistry; Sir George Stokes Medal, Royal Society of Chemistry; Materials Research Society Medal; Silver Medal of the Royal Society for Arts, Manufactures, and Commerce; the American Chemical Society Nobel Laureate Signature Award, and notably, the Acoustical Society of America's Student Council Mentor Award. In 2018, Professor Suslick has been selected as the 76th George Eastman Professor of Oxford University, which is a distinguished Chair appointed annually to a citizen of the United States who is deemed to be of great eminence in teaching or research in any field of study at the University of Oxford. Prior Eastman Professors include 12 Nobel Prize winners.

In addition to his academic research, Professor Suslick has had significant entrepreneurial experience. He was the lead consultant for Molecular Biosystems Inc. and part of the team that commercialized the first echo contrast agent for medical sonography, Albunex™, which became Optison™ by GE Healthcare. These ultrasound contrast agents are, stabilized microbubbles, which have proved to be a major recent contribution to medicine, as the huge scattering cross section of a bubble in resonance permits diagnostic ultrasound scanners to see blood vessels down to the capillary level. In 1990, he invented core-shell microspheres with albumin outer coating and hydrophobic cores using ultrasound for both emulsification and chemical crosslinking of the protein shell. This lead to his role as the founding consultant for VivoRx Pharmaceuticals and the commercialization of Abraxane™, albumin microspheres with a paclitaxel core, which is the predominant delivery system for taxol chemotherapy for breast cancer; VivoRx became Abraxis Bioscience, which was acquired by Celgene for $2.9 billion. He also co-founded ChemSensing and its successors, iSense Systems and Specific Technologies in Mountain View, for the commercialization of optoelectronic nose technology with particular focus on biomedical applications of this unique sensor technology.

Ken has practically invented an entire field of science, namely, that of "sonochemistry." When a liquid, say, water is subjected to an acoustic field of sufficient intensity, the liquid literally tears apart during the rarefaction cycle. The cavity that is formed rapidly grows, filling with vapor, until the compression cycle, and since the vapor can't supply any resistance beyond its vapor pressure, the cavity collapses. When the cavity collapses, any remaining non-condensable gas is subjected to enormous pressures and temperatures. This process is called acoustic cavitation and is the principal basis of sonochemistry. Ken's papers in the early 1980's were instrumental in opening the diversity of cavitation driven chemistry. Sonochemistry as a field has expanded greatly over the past twenty years with a dedicated journal (of which Ken was a co-founding editor) that publishes hundreds of papers annually. One of the consequences of cavitation collapse is that the gas that is compressed is heated to incandescent temperatures—often higher than that of the surface of the sun. This phenomenon is responsible for sonochemical reactions, and also for the emission of light, called sonoluminescence. Ken's articles on sonoluminescence have made the covers of *Science* (twice) and of *Nature*. At one time, over 1,000 papers were published on the topic in a single year, and Hollywood even made a movie (*Chain Reaction*) about it starring

Morgan Freeman and Keanu Reeves, to which Ken and his grad students served as uncredited consultants.

Ken's ingenuity was displayed by his use of cavitation to create completely unique materials. If cavitation is performed in a volatile metallic liquid, the high temperature within the compressed bubble breaks up the molecules into their constituent atoms, and as the bubble cools, the atoms can condense into nanoparticles. Since the cooling rates are greater than 10^{10} Kelvins/second, unique particles are formed, such as amorphous iron or novel metal alloys—which can't be created using conventional means. For this work, and similar research, he was awarded the Materials Research Society Medal.

Ken is not only a brilliant scientist but also an amateur artist and an art collector. Having a special interest in the human face, Ken has created numerous bronze casts of abstracted heads (in the art foundry at the University of Illinois) that display his unique creativity. Of special interest to Ken are ethnographic masks, with a collection of over 400 from all over the world, displayed on the walls of their house and his office: his wife says that there is hardly any wall space left! Ken's mother was an artist and Ken collected his first mask when he was eight, and he still has it. Ken is also known for his memory of lyrics from folk music, Tom Lehrer, and Gilbert & Sullivan (as well as bawdy songs and ballads) and has been an active host to house concerts by musicians passing through central Illinois.

Ken was born in Chicago, eldest of four children of Alvin (M.D.) and Edee (R.N.). He grew up in Glencoe, IL, a suburb on the north shore of Chicago. Ken's son, Benjamin, is also a Caltech alum and is currently pursuing his Ph.D. (in chemistry) at U.C. Berkeley. Ken's wife, Patricia, is a retired school librarian and has previously taught at levels ranging from elementary school up to college freshmen. They have two cats, one nice to Ken and one not so nice.

LAWRENCE A. CRUM
ANDREA PROSPERETTI
RONALD A. ROY

ACOUSTICAL SOCIETY OF AMERICA
GOLD MEDAL

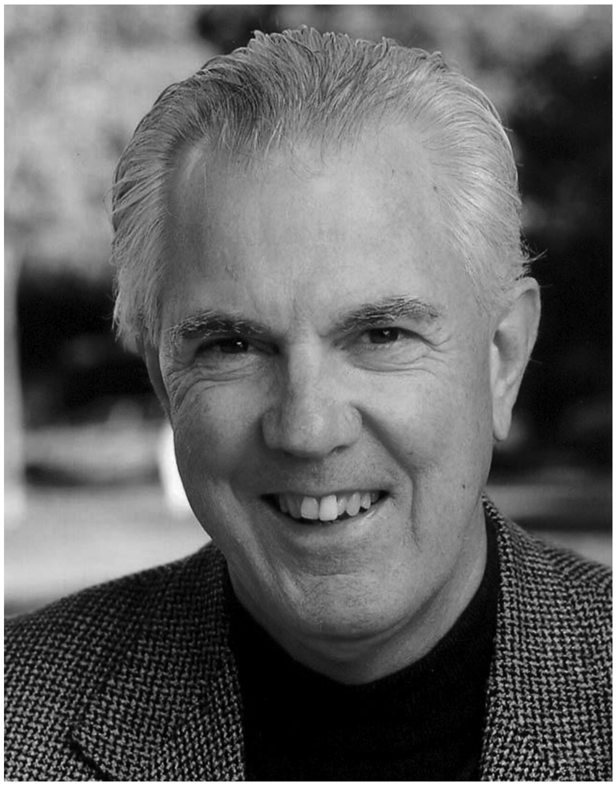
William A. Yost

2018

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



CITATION FOR WILLIAM A. YOST

. . . for research on binaural hearing, pitch and modulation perception and for service to the acoustics community

MINNEAPOLIS, MINNESOTA • 9 MAY 2018

William Albert Yost (Bill) was born in 1944 and grew up in Colorado. He earned his undergraduate degree from Colorado College (1966) where he studied psychology and mathematics and played varsity basketball and tennis. Colorado College would later recognize him with an honorary degree in 1997. The combination of psychology and mathematics took Bill to Indiana University–one of a few American centers where psychologists were transforming one corner of their field into something like a hard science through signal detection theory. During his Indiana days, Bill's romance with Lee began to bloom. They married in 1969 and have two daughters and four grandchildren. Bill received the Ph.D. in Experimental Psychology in 1970 and, like so many others of his generation, began a postdoctoral fellowship with David Green at the University of California, San Diego.

Bill's postdoc time was productive but brief. In 1971 he joined the faculty at the University of Florida–jointly in communication science and in psychology. In 1977 he moved to Loyola University of Chicago where he remained for the next 30 years. At Loyola he served as professor of hearing science, professor of psychology, and director of the Parmly Hearing Research Institute. Later he served as associate vice president for research and dean of the graduate school. In 2007, after Bill and Lee decided that they had seen enough bad weather, Bill became chair and professor of hearing science at Arizona State University where he is now Research Professor.

Throughout Bill's long and varied career, there has been one constant, namely psychoacoustics research. His CV lists more than 100 refereed publications, 67 of them in the *Journal of the Acoustical Society of America* (JASA)... and counting. Bill's training as a student and postdoc prepared him for research in theory and experiment, especially as applied to binaural hearing. He thoroughly absorbed the signal detection techniques, as evidenced by papers from early in his career on the binaural masking level difference. Subsequently, he moved in directions of more directly measurable perceptions, namely binaural sound localization/lateralization and pitch perception.

In the binaural area, Bill did very important experiments associating interaural differences in signal arrival time and signal level with perceived lateralization within the head. Such measurements are notoriously difficult because of biases. It is much easier to do an experiment that asks the listener to identify a change in laterality than to identify laterality absolutely. Not surprisingly, his 1981 paper on the lateral position of sinusoids is his most widely cited binaural research paper. Bill also wrote, together with Ruth Litovsky, Steve Colburn, and Sandy Guzman, the widely cited JASA review of the binaural effect known as the "precedence effect." Overall, Bill's work on localization/lateralization was creative, deep, and impactful, and it, together with his work on the Franssen effect and cocktail party effect, became the cornerstone of his many talks for a general audience and popularizations. Recently, Bill has developed a novel facility to study auditory spatial perception when the source of sound is rotating, or the listener is rotating, or both are independently rotating. This fundamental research has obvious importance for our abilities to understand the world around us and may ultimately be clinically helpful for impaired observers. This extension of binaural research is also fun to think about.

Bill's contributions to the science of pitch have been on the perception of rippled noise. Rippled noise is the result of adding a delayed version of a noise to the original noise itself. The effect is not new. The pitch of rippled noise was discovered by Huygens in 1693 and modern experiments began in The Netherlands in the late 1960s. Bill entered the field in the 1970s struggling with the problem that occurs when the noise is added back with a 180-degree phase shift. That stimulus gives rise to two different, but rather similar, pitches and neither directly reflects the delay. He would go on to write five more articles on rippled noise pitch, pitch strength, and modeling. Then in 1993 Bill discovered iterated rippled noise, where the delay-and-add process for rippled noise is repeated to make finite or infinite impulse response filters. That variation led to 11 more publications, and these are Bill's most widely cited research articles.

Underlying the interest in rippled noise pitch is the hope that the combination of theory and experiments will clarify the origin of pitch itself. Thus, although Bill did not invent rippled noise, he brought the study to the United States and promoted it at every opportunity. As a result, rippled noise became a standard stimulus. Articles in JASA show it applied to hearing impairment, vertical plane localization, infant screening, auditory filter measurement, cochlear implants, chinchillas, budgerigars, bottlenose dolphins, and beluga whales.

Starting in the 1980s Bill did important early work on modulation detection interference (MDI), including the important step of adding interaural differences. MDI is observed when a slow amplitude modulation in one signal or noise (called the masker) interferes with the ability to detect amplitude modulation in a different signal or noise. The effect occurs even if the two signals are separated by several octaves. Interference is greatest when the masker has the same modulation frequency as the signal. The unimportance of carrier frequency and the tuning in modulation clearly emerged from Bill's work. The problem with MDI is to understand its origins. It is known that there are tuned modulation channels in the auditory system, and MDI may reflect nothing more than masking within these channels. Alternatively, MDI may have a central origin, reflecting the ability of common modulation to fuse signals perceptually, such as two tones in very different frequency ranges.

From the earliest days of his academic career Bill has contributed to education in hearing science. His undergraduate textbook *Fundamentals of Hearing*, originally co-authored with Donald Nielsen in 1977 and now in its 5th edition, has introduced hearing science to hundreds of thousands of students. He is the author of five other books and 45 book chapters or conference proceedings. Over the years Bill has mentored junior colleagues who remain active in research and in the ASA. Colleagues find that Bill's excellent memory for facts and ideas in psychoacoustics is a valuable resource.

Everywhere he goes Bill gets involved with what is going on. His record shows contributions to noise standards for forensic scientists, noise ordinances for Florida, chairing committees for the National Research Council, lectures on computers in education, and letters to *Science*.

Bill gave a talk at the Cleveland meeting of the ASA in November 1968, and he has been active in the Society ever since. He has held every elective office that the Society has available, serving as president in 2005. He was the ASA co-chair of the largest acoustics meeting in history (ASA-EAA Paris, 2008). Although he has been active as a member and an officer of other scientific organizations, the ASA has been his main intellectual home, and the ASA has benefited from his administrative skills. Bill is the consummate committee chair. He has an uncanny ability to read the sense of the room and to guide discussions to successful conclusions.

Because of Bill's talents as an administrator, a convener, and a leader there was really no limit to how far he might have gone in university or government administration. With his background and personality he could have assumed high profile roles elsewhere if he had wanted to do it. Fortunately for us, he remained dedicated to research in psychoacoustics, and we find it most appropriate that the Society awards him its Gold Medal.

WILLIAM M. HARTMANN
ROY D. PATTERSON
H. STEVEN COLBURN

Session 4aAAa

Architectural Acoustics, Noise, ASA Committee on Standards, and Speech Communication: Acoustics in Classrooms and Other Educational Spaces

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

Chair's Introduction—7:55

Invited Papers

8:00

4aAAa1. China green campus school standard. Kenneth P. Roy (Innovation, Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17603, kproy@armstrongceilings.com) and Xiao Li (Res. and Development, Armstrong World Industries, Shanghai, Shanghai, China)

Since 2011 I've been involved as a principal investigator (Armstrong World Industries) on several research programs encompassing approximately 100 primary and middle grade schools throughout China. Initially, this research was coordinated with Tongji University for the development of the Chinese Green Campus rating system for schools, and later, we have worked with the Beijing Ministry of Education and several additional Universities including Tsinghua Univ., South China Univ. of Technology, Chongqing Univ., and Huaqiao University for the development of the new China GB Green Campus Standard. The "learnings" from these research studies will be reviewed, and the outcomes in terms of implementation will be discussed. A "playbook" was developed for the government architects who design schools, as a simple process for meeting the design goals, and understanding the installed costs for specific use areas. School acoustics is an important IEQ factor in the design and performance of the school in meeting its mission—which is for teachers to teach and students to learn.

8:20

4aAAa2. A review of acoustic research in physical educational settings. Stuart Ryan (Univ. of West Florida, 11000 University Pkwy, Pensacola, FL 32514, sryan@uwf.edu)

This presentation will review past and current acoustic research in physical education settings. Research suggests that the acoustic issues in physical education settings are likely detrimental to student learning (Ryan, 2009a; Ryan, Grube, & Mokgwati, 2010; Ryan & Mendell, 2010). The problem is compounded if there are students with special needs including learning disabilities, attention and auditory disorders, and students who are listening and learning in a non-native language. In addition, typical classroom teachers are faced with many voice concerns; however, physical education teachers are faced with poor acoustically designed gymnasiums, covered areas, and loud outdoor teaching environments which are more challenging than a typical indoor classroom (Ryan, 2009a). Ryan (2010) viewed physical education settings as "hostile listening environments" and have the potential to effect student learning and damaging teacher's voices.

Contributed Papers

8:40

4aAAa3. Acoustical analysis of preschool classrooms. Tina M. Grieco-Calub (Roxelyn and Richard Pepper Dept. of Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., FS, 2-231, Evanston, IL 60208, tinagc@northwestern.edu) and Z. Ellen Peng (Waisman Ctr., Univ. of Wisconsin-Madison, Wisconsin, United States Minor Outlying Islands)

The present study explores acoustical parameters, including unoccupied and occupied noise levels and reverberation time (RT), in typical preschool classrooms located in the northern suburbs of Chicago. The study was motivated by the following observations: (1) preschool classrooms are often established in buildings that were not initially constructed to be learning spaces; (2) poor classroom acoustics interfere with skills related to academic outcomes including speech perception, serial recall, literacy, and cognitive skills; and (3) younger children are at greater risk for impaired performance in acoustically complex environments. The study was designed to determine whether these preschool classrooms meet existing classroom acoustics standards (ANSI S12.60-2010). Measurements were made with microphones positioned in locations of the classrooms where children typically engage in activities during school season to reflect realistic building operation situations. Unoccupied and occupied noises were recorded over long durations, with intensity and spectral analysis of the recordings conducted off-line. RTs were measured and quantified at the same microphone positions. Preliminary results suggested that none of the preschool classrooms met the recommended unoccupied noise levels, while 3/11 classrooms were found to have longer than recommended RTs. Occupied noise in these preschool classrooms, and implications for learning, will be discussed.

8:55

4aAAa4. Classroom soundscape—Virtual and real. Andy Chung (Smart City Maker, Hong Kong Plaza, Hong Kong HKSAR, Hong Kong, ac@smartcitymaker.com) and W M To (Macao Polytechnic Inst., Macao, Macao)

The sonic environment which both teachers and students are provided in a classroom is important for effective communication during the learning process. How to design and optimize for a classroom requires a study of its soundscape including the feedback of potential users on their perception of the sonic environment. This paper presents how virtual technology is used to characterize the classroom soundscape and how it is related to the reality.

9:10

4aAAa5. Noise in and around schools. Sergio Beristain (IMA, ESIME, IPN, P.O.Box 12-1022, Narvarte, Mexico City 03001, Mexico, sberista@hotmail.com)

Incoming noise is usually a major problem for many schools which can drastically affect the teaching process, and often times this is the case due mainly to the traffic and community noise, but also in many cases the schools become themselves large noise sources, affecting the community tranquility. A case study is presented which shows the noise levels and acoustics characteristics of a basic school located in the middle of a middle class suburb, emphasizing the main nuisances that the school receives from the surroundings, and also those produced by the school to the community throughout its normal activities from before opening the school doors in the morning, till the last student has leaved, and during the teaching and leisure activities of young students.

9:25

4aAAa6. Multifunction speech intelligibility in a renovated historic room. Sarah L. Bochat, Michael B. Doing, Peggy B. Nelson (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr SE, Minneapolis, MN 55455, bocha007@umn.edu), Ana M. Jaramillo (AFMG Services North America, LLC, Brooklyn Park, MN), and Bruce Olson (Olson Sound Design, Brooklyn Park, MN)

Shevlin Hall, a historic building erected in 1906, is presently home to the department of Speech-Language-Hearing Sciences on the University of Minnesota campus. The primary classroom space is a 20' x 50' room with a high coffered ceiling used for lecture-style classes, discussion groups, and social events. The space underwent renovation in 2013 to address room acoustics and sound design. Post-renovation acoustic measurements and subjective responses reflect favorable outcomes for lecture conditions. Challenges remain for small-group and active discussion classroom arrangements. The current study evaluated predicted and measured speech intelligibility for small-group and active discussions. Preliminary data suggest good intelligibility throughout the room for the condition of lecture-style presentation. Reduced intelligibility is suggested for the condition of talker in back and receiver at instructor or front center position. Implications for sound design will be discussed.

9:40

4aAAa7. Exploring relationships between measured acoustic, lighting, and indoor air quality metrics logged in K-12 classrooms. Kieren H. Smith and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, PKI 210C, 1110 S. 67th St., Omaha, NE 68104, kierenhs@gmail.com)

Comfort in classrooms engenders learning and countless factors contribute to the quality of an educational environment. A longitudinal study measuring classroom conditions, including indoor air quality, lighting, and acoustic components has amassed data from 220 schools in the Eastern Nebraska area. Measurements logged at time intervals no longer than 5 minutes were taken for two full days in each classroom, three times in a year. Using this data, the relationships over time that acoustic factors have with indoor air quality, thermal comfort, and lighting factors are explored. Correlations are investigated between average measured values, time and spatial variation of measured values, and spectral composition of the acoustic metrics. Besides correlations between the assorted measured quantities, other relationships between the design of the building mechanical and lighting systems and resulting acoustic conditions are discussed; for example, do heat pumps or VAV boxes contribute to significantly different acoustic conditions? [Work supported by the United States Environmental Protection Agency Grant No. R835633.]

Session 4aAAb

Architectural Acoustics: Topics in Architectural Acoustics—Measurements, Perceptions, and Metrics

Ian B. Hoffman, Cochair
Peabody Institute: Johns Hopkins Univ., 1 E Mount Vernon Place, Baltimore, MD 21202

Shane J. Kanter, Cochair
Threshold Acoustics, 53 W. Jackson Blvd., Suite 815, Chicago, IL 60604

Chair's Introduction—10:10

Contributed Papers

10:15

4aAAb1. Noise level prediction in a bakery using panel contribution analysis combined with scale modeling. Gong Cheng and David Herrin (Dept. of Mech. Eng., Univ. of Kentucky, 151 Ralph G. Anderson Bldg., Lexington, KY 40506-0503, gongcheng.vac@uky.edu)

Panel contribution analysis combined with scale modeling was utilized to determine the sound pressure level at different positions in a bakery. The primary sources of noise were three heating and air conditioning units. Each of the units was discretized into several patches and the volume velocity for each patch was measured. Transfer functions were measured between the patches and two different customer locations for the full-scale baseline case and using a 1/10 scale model. Transfer functions for the scale model were adjusted to account for air attenuation in the full-scale case. Panel contribution analysis was used to predict noise levels assuming both correlated and uncorrelated sources. Predicted results compared well with measurement. The suggested method can be used to assess the sound pressure levels in large interior spaces before they are constructed so long as the source positions are known. In addition, contributions from the individual sources can be assessed and ranked.

10:30

4aAAb2. Analyzing frequency responses from time-windowed impulse responses across assorted types of room usage. Brenna N. Boyd and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 137 N Jefferson St., Apt A, Papillion, NE 68046, bnboyd@unomaha.edu)

Lokki *et al*. (2015) examined how frequency responses in concert halls change over different time-windowed periods of the impulse response across low, mid, and high frequencies, and connected that with perceived quality in those halls. Time windows were applied to highlight the direct, early, late, or total sound energy, and the resulting frequency responses were overlaid on the same graph to view easily the difference and growth between the time periods across frequencies. In this project, such frequency response analyses across different time-windowed impulse responses are applied to spaces other than concert halls, including a variety of classrooms, worship spaces, and music-centered and speech-centered performance spaces. The objective is to explore the utility of this method of analysis for assorted room usages. In this presentation, interesting cases and trends within individual rooms, within one type of room usage, and between different types of room usage are discussed, as well as this analysis method's advantages and disadvantages.

10:45

4aAAb3. New research on low-frequency membrane absorbers. John Calder (Acoust. Surfaces, Inc., 123 Columbia Court North, Chaska, MN 55318, jcalder@acousticalsurfaces.com)

Low-frequency (LF) room modes are one of the greatest issues for accurate sound recording and reproduction. Effective LF absorbers can mitigate modes in professional and consumer audio rooms. However, fiber- and foam-based absorbers act on sound *velocity*; membrane absorbers act on sound *pressure* (greatest at hard surfaces and corners). Velocity at hard surfaces is zero; thus, fiber and foam absorbers work far less effectively than membrane absorbers under 200Hz. Additionally, most independent testing laboratories are only large enough to accurately measure absorption results above 160Hz (per Schroeder frequency) but not below. Only one lab is large enough to be accurate down to 40Hz. A new LF membrane-based absorber product was designed to compliment the frequency range of an existing product. Both were separately tested for LF absorption down to 40Hz at the above-referenced lab. Ten LF absorber tests revealed that the type of absorber, and its location and orientation in a room, are critical to LF absorber effectiveness. Some unexpected results, however, showed clearly that without standardized laboratory absorption testing in a lab capable of accurately testing down to 40Hz, it is not possible to state conclusively that low-frequency absorber products perform as claimed.

11:00

4aAAb4. A research about influences to piano players' perception and behavior control in brushing up processes of music pieces under the condition of different reverberation time sonic fields. Ayako Matsuo and Takeshi Akita (Sci. and Technol. for Future Life, Tokyo Denki Univ., 5 Asahi-cho Senju Adachi-ku, Tokyo 120-8551, Japan, matsuoaya.dendai@gmail.com)

Dividing piano players' proficiency processes of music pieces into three categories of early, middle and final stages, we have studied about the difference and characteristics of their perception for sonic environment and control of performance behaviour at each stage.In our previous studies, it has been shown that we could measure the difference and characteristics of their "perception and behaviour control" on performance mainly at the early and the final stages in brain activity, using NIRS (Near-infrared spectroscopy) method and having some supplementary interviews after the experiments.Moreover in this paper, we carried out experiments about players' "perception and behavior control" on performance at the two stages under some different reverberation time sonic fields using same method.We also

4a THU. AM

made sure of the cognition of the players' performances by having some supplementary interviews after the performance experiments.As a result, we could obtain that players' "perception and behavior control" on performance is being influenced from the condition of reverberation time difference of sonic field.

11:15

4aAAb5. Amplified music and variable acoustics—Controlled low frequencies and envelopment at high. Niels W. Adelman-Larsen (Flex Acoust., Diplomvej 377, Kgs. Lyngby 2800, Denmark, nwl@flexac.com)

Subjective ratings from 25 professional musicians and sound engineers were obtained to assess two Danish rock venues of similar size and similar low frequency reverberation times, but different high frequency reverberation characteristics. The musicians rated one hall outstanding and significantly better than the other, substantiating a hypothesis that pop venues can have a longer reverberation time at mid to high frequencies at least in the empty condition with no seats installed. Possible causes for this is an enhancement of the musicians' ease of expressing dynamics as well as of their ability to sense the audience for whom they are playing. It has formerly been shown, that what characterizes the best halls for amplified music is a relatively short reverberation time at low frequencies. However, in this study, a fairly long reverberation time in the 63 Hz octave band is found to be acceptable, especially in the hall with the longer mid-high frequency reverberation time, pointing towards a tolerance for longer reverberation in this band in general. Further, this hints at the possibility that mid-high frequency reverberation can mask, at least 63 Hz band challenges. The 125 Hz octave band RT, which thereby remains the single most important to control for amplified music, is app. 1 sec in the two venues that hold some 700 standing audiences

11:30

4aAAb6. On the prediction of sound diffusion coefficient. Hassan Azad and Gary W. Siebein (Architecture, Univ. of Florida, 3527 SW, 20th Ave, 1132B, Gainesville, FL 32607, h.azad@ufl.edu)

Acoustic diffusers are one of the key elements in architectural acoustic design. The shape, placement and combination with absorbers along with the physical characteristics of the room such as the volume, size, and geometry make a complex design scenario full of variables and dependencies. The Prediction of scattered reflections with the help of computer modeling is one of the most recent areas of research. Currently the only software package that exist for the simulation of the scattered sound is Reflex [1] from AFMG which can predict scattering and diffusion coefficients of 1D acoustics diffusers using their cross sections. It employs Boundary Element Methods for the calculation purposes. This study intends to develop a sound diffusion prediction software that simulates the measurement laboratory environment of the diffusion coefficients in accordance with 17497-2 ISO standards to predict diffusion coefficients of different type of 2D and 3D acoustics diffusers such as Schroeder, fractal and volumetric diffusers. For the calculations, wave-based room acoustics modeling techniques such as FDTD and BEM will be used. There are plenty of partial differential solvers available that are used for acoustics purposes which will be used in this paper as well. [1] http://reflex.afmg.eu/

11:45

4aAAb7. First-order approximation of diffraction at a rectangular rigid plate using the physical theory of diffraction. Aleksandra Rozynova (Acentech, 33 Moulton St., Cambridge, MA 02138, arozynova@acentech.com) and Ning Xiang (Rensselaer Polytechnic Inst., Graduate Program in Architectural Acoust., Troy, NY)

Efficient predictions of sound diffraction around objects is vital in room-acoustics simulations. Recent efforts using the Physical Theory of Diffraction (PTD) [Rozynova, A. and Xiang, N., *J. Acoust. Soc. Am.*, **141**, pp. 3785 (2017)] have been reported on some canonical objects of infinite size. This research has been further developed based on solving diffraction problems on finite, rectangular plates which are free from grazing singularities. Separate solutions are considered for diffraction from two pairs of the rigid plate edges, the edge waves around the plate corner are ignored due to the lower order of diffraction contributions in the far-field. This paper will discuss numerical implementations of the theoretical predictions to demonstrate the efficiency of the PTD in room-acoustics simulations for finite sized rectangular plates. The implemented numerical results will also be compared with some preliminary experimental results carried out with a goniometer.

Session 4aAB

Animal Bioacoustics: Animal Sound Production and Hearing

Micheal L. Dent, Chair
Psychology, University at Buffalo, SUNY, B76 Park Hall, Buffalo, NY 14260

Chair's Introduction—8:00

Contributed Papers

8:05

4aAB1. Temporal fine structure and zebra finch vocalizations. Nora H. Prior, Edward Smith, Gregory F. Ball, and Robert Dooling (Psych., University of Maryland, 4094 Campus Dr, College Park, MD 20742, nhprior@umd.edu)

The temporal fine structure (TFS) of an acoustic signal refers to the smaller changes within the time waveform, independent of larger envelope changes. Bioacousticians typically have relied on sonographic analyses to describe the primary acoustic features of animal vocalizations and human speech. However, sonographic analyses do not reveal the temporal fine structure in complex vocal signals. Recent research on the perception of human speech by normal-hearing and hearing-impaired listeners has suggested that TFS is important for speech comprehension. Interestingly, birds have much better sensitivity to changes in TFS than do humans, suggesting TFS could also be important for communication in birds. TFS in complex signals is extremely difficult to study in isolation since it occurs along with other acoustic features of complex signals. But, here using a variety of techniques to reduce or eliminate other acoustic features of a complex acoustic signal, we show that zebra finches have particularly good perceptual sensitivity to the natural variation in TFS, and that there are very likely systematic differences in TFS that are unique to various classes of vocal signals. These results suggest that, as with human speech, TFS may be important in avian acoustic communication.

8:20

4aAB2. Calls of the bald eagle (*Haliaeetus leucocephalus*). Edward J. Walsh (Res., Boys Town Natl. Res. Hospital, 555 North 30th St., Omaha, NE 68132, edward.walsh@boystown.org), Peggy Nelson, Andrew Byrne, Jeff Marr, Christopher Feist, Christopher Milliren, Julia Ponder, Patrick Redig (Univ. of Minnesota, Minneapolis, MN), and JoAnn McGee (Res., Boys Town Natl. Res. Hospital, Omaha, NE)

Injury and mortality statistics suggest that bald eagles entering the air space of wind energy facilities face considerable risk. To mitigate the hazard, acoustic deterrent systems designed to discourage entry into such hazardous air spaces are under consideration. In this study, the acoustic properties of a collection of call types within the eagle vocal repertoire are reported as a first step in a larger program of study designed to assess the deterrent capacity of the bird's natural vocal utterances. To that end, calls were recorded from bald eagles housed in the Raptor Center at the University of Minnesota. Based on preliminary acoustic analyses, at least five calls were identified and are referred to here as peal/scream, chatter/cackle, snort, squeal and grunt. With the exception of the low frequency grunt, calls were uniformly high pitched, tonal in nature, exhibited harmonic spectral structure, and they were generally complex, exhibiting distinct nonlinear characteristics. In this presentation, the spectrotemporal properties of each call type will be described with the goal of generating an acoustic repository to enable the consistent classification of calls to be tested for their potential as deterrence signals. [This work was supported by Department of Energy grant #DE-EE0007881.]

8:35

4aAB3. Use of transition and abstract rules by budgerigars in auditory pattern processing. Adam Fishbein, Gregory F. Ball, and Robert Dooling (Neurosci. & Cognit. Sci., Univ. of Maryland, College Park, 4123 Biology-Psych. Bldg., University of Maryland, College Park, MD 20742, afishbei@umd.edu)

Budgerigars, a small species of parrot, are open-ended vocal learners that produce a long, rambling warble song. Past work has shown that these birds are sensitive to changes in the order of their song elements but it is not clear what rules they may use to process changes in sounds sequences. Here, we used operant conditioning and psychophysical techniques to ask how budgerigars discriminate among patterned auditory sequences. In these experiments, auditory sequences of the form AAB, composed of pure tones or natural warble elements, were played to the bird as a repeating background until the bird pecked a key to initiate a trial. During a trial, either a target sequence played that differed from the background in the order of the elements or a sham trial occurred where there was no change. Results show that budgerigars use rules about the transitions between elements to detect targets. Further experiments using multiple background sequences show that the birds are also sensitive to the underlying abstract pattern of sounds (same-same-different). Thus, with their ability to attend to surface transition rules (phonology-like) and deeper abstract patterns (syntax-like), budgerigars are a useful vocal learning model for studying the neurobiology and evolution of language.

8:50

4aAB4. Ultrasonic vocalization production by socially isolated and experienced mice. Laurel A. Screven and Micheal L. Dent (Psych., Univ. at Buffalo, B80 Park Hall, Buffalo, NY 14260, laurelsc@buffalo.edu)

Mice produce ultrasonic vocalizations (USVs) in a wide variety of social situations, including courtship, investigation, and territorial defense. Although production of USVs is believed to be innate, suggesting mice do not need acoustic experience with conspecifics to produce calls, it is possible mice require experience to emit calls in the proper behavioral context. Male mice require social experience with a female to produce USVs in response to olfactory signals present in dirty bedding (e.g., Nyby *et al.*, 1983), and isolated male mice will produce more USVs than socially experienced males (Keesom *et al.*, 2017), however the influence of social experience on vocal production by female mice has yet to be investigated. The present experiment aimed to determine if social isolation or experience with conspecifics influences the vocal repertoire of adult female CBA/CaJ mice. Mouse calls were recorded during exposure to an unknown male or female mouse and analyzed to determine if isolation or experience affected call number, rate, or spectrotemporal complexity. Chronic social isolation did influence the vocal repertoire, suggesting that social experience plays a role in the production of USVs by female mice.

9:05

4aAB5. The effects of noise and age on the detection of ultrasonic vocalizations and pure tones by laboratory mice. Anastasiya Kobrina and Micheal L. Dent (Psych., SUNY Univ. at Buffalo, B23 Park Hall, SUNY University at Buffalo, Amherst, NY 14261, akobrina@buffalo.edu)

Mice are frequently used to study and model age-related hearing loss (ARHL) due to similarities in human and mouse cochleae and in genetic makeup. Mice emit ultrasonic vocalizations (USVs) that may be used for communication. Behavioral research has established that mice exhibit ARHL for detection of USVs and pure tones, however later in life than previously estimated by electrophysiological methods. In addition, mice showed sex-related ARHL differences for USVs but not for 42 kHz tones. In order to fully understand whether mice can be used to model ARHL it is critical to examine hearing loss for simple and complex stimuli in a variety of listening environments. In this study, detection thresholds were measured in adult mice using operant conditioning procedures and positive reinforcement. Mice were trained to detect USVs and high frequency pure tones in silence and in 50 dB white noise. The results revealed that adult mice retain hearing late into their lifespan, with very old mice exhibiting higher thresholds for USVs and pure tones in noise than in silence. The results highlight the importance of measuring hearing in awake, trained, behaving subjects when comparing ARHL across different species. [This work was supported by F31 DC016545 and R01 DC012302.]

9:20

4aAB6. The use of background noise to mask objectionable sounds to rodents. Robert D. Bruce, Arno S. Bommer, Isaac Harwell, Adam Young, and Max Meynig (CSTI Acoust., 16155 Park Row # 150, Houston, TX 77084-6971, bob@cstiacoustics.com)

Construction noise and vibration can interrupt sleep/concentration of humans during renovations. Noise treatments or vibration breaks can reduce the impact. But could background noise cover up a moderate level of construction noise? Innocuous sounds are used to mask quieter but more annoying sounds. The concept is commonly used in offices to aid in concentration and reduce annoyance in places where people work or sleep. But could it be used to mask construction noise if the recipients were rodents rather than people? In theory, the concept of sound masking should be valid for other species of animals. Excessive noise and vibration has been shown to affect rodent behavior including breeding. Their hearing, extending into the ultrasound region, complicates efforts to recognize and measure the rodents' acoustical environment. During the past several years, we have had occasion to study the literature on rodents' hearing and the use of masking to prevent the creatures from being bothered by intruding sound. In this paper, we summarize the literature and investigate the concept of masking for rodents with respect to regularly encountered sound sources and sounds from construction activities.

9:35

4aAB7. Optical coherence tomography reveals complex motion between the basilar membrane and organ of corti in the gerbil cochlea. Clark E. Strimbu (Dept. of Otolaryngol., Head and Neck Surgery, Columbia Univ. Medical Ctr., P&S 11-452, 630 West 168 St., New York, NY 10032, ces2243@cumc.columbia.edu), Nathan C. Lin (Dept. of Elec. Eng., Columbia Univ., New York, NY), and Elizabeth S. Olson (Dept. of Otolaryngol., Head and Neck Surgery, Columbia Univ. Medical Ctr., New York, NY)

The mechanical basis of auditory transduction within the mammalian cochlea is not fully understood but relies on complex mechanical interactions between different structures within the Organ of Corti (OoC). We used spectral domain phase microscopy, a functional extension of optical coherence tomography (OCT), to measure the sound-evoked motion in response to "Zwuis" frequency complexes, *in vivo*, in the basal region of the gerbil cochlea. Because OCT allows for measurements from multiple axial positions along the optical path, we are able to simultaneously record vibration at different points within the OoC between the basilar membrane (BM) and tectorial membrane. Imaging through the round window membrane, we could record from several longitudinal locations along the cochlea and at each location, we measured from several radial positions. Both the BM and intra-OoC vibrations are tuned and show nonlinear amplification, or compressive growth, with the applied sound pressure level. Similar to recent reports (Ren *et al*, PNAS 2016; Ren and He, MoH 2017), we find that the motions of surfaces within the OoC can exhibit a greater degree of compressive nonlinearity (amplification) and physiological vulnerability than the motion of the BM. Additionally, some responses show sharp notches near the onset of the nonlinear regime.

9:50

4aAB8. Copulation calls of female chimpanzees (Pan troglodytes) signal fertility. Anthony P. Massaro (Ecology, Evolution, and Behavior, Univ. of Minnesota, 1987 Upper Buford Cir, Falcon Heights, MN 55108, massa060@umn.edu), Emily E. Boehm (Evolutionary Anthropology, Duke, Durham, NC), Elizabeth V. Lonsdorf (Psych. and Biological Foundations of Behavior Program, Franklin & Marshall College, Lancaster, PA), Carson M. Murray, Margaret A. Stanton (Ctr. for the Adv. Study of Human Paleobiology, The George Washington Univ., Washington DC, DC), and Michael Wilson (Anthropology, Univ. of Minnesota, Minneapolis, MN)

In many species, females produce vocalizations during or soon after mating. In chimpanzees, these calls accompany only some matings while visual signals of sexual receptivity are always visible. Proposed explanations for chimpanzee copulation calls include (1) increasing paternity confusion and thus reducing infanticide risk, and (2) competing more effectively with other females for male mating interest. We used information theoretic model selection to test competing hypotheses using a record of 1,168 copulation events recorded by the mother-infant project team at Gombe National Park, Tanzania (1976-2015). We found that females produced copulation calls more frequently in the last week of their reproductive cycle, when they are most fertile. This timing is consistent with the hypothesis that copulation calls function as an honest signal of fertility. While females also signal fertility visually (with sexual swellings), copulation calls may signal to a broader audience, encouraging males to invest in mating effort. Male primates are often portrayed as always being ready to mate, but in reality, mating entails potential costs for males, including the risk of aggression from other males, and opportunity costs that must be weighed against the risk of mating with a non-ovulating female.

10:05–10:20 Break

10:20

4aAB9. Vessel sound cuts down communication space for vocalising fish and marine mammals. Rosalyn L. Putland (Leigh Marine Lab., Inst. of Marine Sci., Univ. of Auckland, 160 Goat Island Rd., Leigh, Auckland 0985, New Zealand, rput037@aucklanduni.ac.nz), Nathan D. Merchant, Adrian Farcas (Ctr. for Environment, Fisheries and Aquaculture Sci., Lowestoft, Suffolk, United Kingdom), and Craig A. Radford (Leigh Marine Lab., Inst. of Marine Sci., Univ. of Auckland, Warkworth, New Zealand)

Anthropogenic noise across the world's oceans threatens the ability of vocalizing marine species to communicate. Some species vocalize at key life stages or whilst foraging, and disruption to the acoustic habitat at these times could lead to adverse consequences at the population level. To explore the risk of such impacts, we investigated the effect of vessel noise on the communication space of the Bryde's whale *Balaenoptera edeni*, an endangered species which vocalizes at low frequencies and bigeye *Pempheris adspersa*, a nocturnal fish species which uses contact calls to maintain group cohesion while foraging. By combining long-term acoustic monitoring data with AIS vessel-tracking data and acoustic propagation modelling, a quantitative method for determining the impact of vessel noise on their communication space was established. Routine vessel passages cut down communication space by up to 61.5% for bigeyes and 87.4% for Bryde's whales. The influence of vessel noise on communication space also exceeded natural variability between 3.9 and 18.9% of the monitoring period. To combat potential effects of vessel sound, we propose the application or extension of ship speed restrictions in ecologically significant areas, since our results indicate a reduction in sound source levels for vessels transiting at lower speeds.

10:35

4aAB10. Analysis of Haddock (*Melanogrammus aeglefinus*) sounds recorded in the Northwest Atlantic. Xavier Mouy (School of Earth and Oceans Sci., Univ. of Victoria, PO Box 1700, Victoria, BC V8W3P6, Canada, Xavier.Mouy@jasco.com), Rodney A. Rountree (Biology, Univ. of Victoria, Waquoit, MA), Katie A. Burchard (Northeast Fisheries Sci. Ctr., NOAA Cooperative Res. Study Fleet Program, Narragansett, RI), Francis Juanes (Biology, Univ. of Victoria, Victoria, BC, Canada), and Stan E. Dosso (School of Earth and Oceans Sci., Univ. of Victoria, Victoria, BC, Canada)

Haddock (*Melanogrammus aeglefinus*) are present on both sides of the North Atlantic. Their distribution in the Northwest Atlantic ranges from Greenland to North Carolina. They are an important food resource that needs to be closely monitored to ensure a sustainable fishery. Research studies have reported that both male and female haddock produce sounds during courtship and spawning. These sounds can be used to monitor spawning activities non-intrusively and at large scale. The objective of this paper is to analyse the spatial and temporal occurrence of Haddock sounds in the Gulf of Maine. Passive acoustic data were collected in 2003, 2004, 2006, and 2007 in areas known to contain spawning haddock. To analyze the large amount of data collected, an automated haddock sound detector was developed based on a measure of kurtosis and the Dynamic Time Warping algorithm. The detector was trained using the 2006 and 2007 data and its performance was quantified and optimized by comparing detection results with manually annotated Haddock sounds. The detector was then used to analyze data collected in 2003 and 2004. Results provide information on the temporal and spatial distributions of courtship and spawning sounds.

10:50

4aAB11. A frog's lungs sharpen the frequency tuning of its peripheral auditory system: A pulmonary paradox resolved? Mark Bee (Dept. of Ecology, Evolution, and Behavior, Univ. of Minnesota, 100 Ecology, 1987 Upper Buford Circle, St. Paul, MN 55108, mbee@umn.edu), Norman Lee (Dept. of Biology, St. Olaf College, Northfield, MN), and Jakob Christensen-Dalsgaard (Biology Inst., Univ. of Southern Denmark, Odense M, Denmark)

In frogs, sound localization is facilitated by the directionality of their internally coupled ears, which are connected through the mouth cavity and Eustachian tubes. Acoustic input through the frog's body wall and lungs also shapes the ear's directionality. One hypothesized function of the lung input is to enhance directional hearing to improve localization of conspecific calls. However, an unresolved paradox exists: the lung input often improves directionality at frequencies that are intermediate between those emphasized in conspecific vocalizations. The present study of green treefrogs (*Hyla cinerea*) may resolve this paradox. Using laser vibrometry, we show that the lung input improves directionality at frequencies emphasized in conspecific calls by just 2 dB or less. In contrast, the lung input reduces the frontal field sensitivity of the tympanum by up to 7 dB in response to frequencies that are intermediate between those emphasized in conspecific calls and to which the two inner ear papillae are tuned. These results suggest the lung input may function in communication by sharpening the frequency tuning of the peripheral auditory system, which in turn could filter out frequencies in heterospecific calls in some mixed-species choruses. How the lung input achieves this "noise cancellation" is currently under investigation.

11:05

4aAB12. Temporal information processing in gray treefrogs: Compute within frequency then integrate across frequency. Saumya Gupta and Mark Bee (Ecology, Evolution and Behavior, Univ. of Minnesota, 100 Ecology, 1987 Upper Buford Circle, St. Paul, MN 55108, gupta333@umn.edu)

In Cope's gray treefrog (*Hyla chrysoscelis*), the pulse rate of advertisement calls is a species recognition cue. This temporal information is redundant across two spectral bands in the call's bimodal spectrum that are primarily transduced by different inner ear papillae. Previous studies have shown that some neurons in the frog auditory midbrain are selective for bimodal spectra and that, compared with unimodal spectra, calls with bimodal spectra are more attractive. In addition, "interval-counting" neurons in the frog auditory midbrain are selective for conspecific pulse rates. However, it is not known whether pulse rate information is computed prior to, or subsequent to, spectral integration. We conducted behavioral experiments to investigate the serial order of pulse rate computations and spectral integration. In phonotaxis tests that used bimodal and unimodal stimuli with variable pulse rates, we tested the hypothesis that spectral information is integrated prior to pulse rate computations versus the alternative hypothesis that pulse rate can be computed independently in different frequency channels prior to spectral integration. Our results support the latter hypothesis, and suggest pulse rate can be computed independently in separate frequency channels. These results give insight into the mechanisms of temporal processing underlying the perception of key biological information.

11:20

4aAB13. Within-individual variation in signal production limits female choice in noisy treefrog choruses. Jessie C. Tanner (Ecology, Evolution, and Behavior, Univ. of Minnesota - Twin Cities, 1479 Gortner Ave, 140 Gortner Labs, Saint Paul, MN 55108, tanne123@umn.edu) and Mark Bee (Ecology, Evolution, and Behavior, Univ. of Minnesota - Twin Cities, St. Paul, MN)

In acoustically communicating animals such as anurans and orthopterans, individual signalers can display remarkable variation in signal traits, even within a single bout of calling. This within-individual variation has the potential to "mask" the between-individual variation that is the target of sexual selection. Thus, receivers may not always be able to discriminate among males based on properties of their acoustic signals, even when female choice is well documented in the lab. The effect of within-individual variation may be compounded by noise generated by breeding choruses of conspecifics, which can prevent signal detection or discrimination. We manipulated within-individual variation in two signal traits (call rate and pulse rate) that are critically important for mate choice in *Hyla chrysoscelis*, Cope's gray treefrog. We replicated our design at three realistic levels of chorus noise. Within-individual variation in signal traits reduced the strength of sexual selection on one of two traits, and receivers were less likely to discriminate against lower-quality signals under conditions of realistic chorus noise. Our results suggest that preference functions generated using stimuli that lack within-individual variation, under ideal listening conditions, may chronically over-estimate the strength of sexual selection on signal traits in natural populations.

11:35

4aAB14. Effect of broadband chirp signal on the sonic behavior of the /kwa/ fishes inhabiting Posidonia oceanica seagrass meadow. Jean-Pierre Hermand and Laura Velardi (LISA - Environ. HydroAcoust. Lab, Université libre de Bruxelles, av. F.D. Roosevelt 50, Brussels 1050, Belgium, lvelardi@ulb.ac.be)

Early experiment to study acoustics of Posidonia oceanica seagrasses (USTICA99) transmitted acoustic chirp signals (0.2-16 kHz, 15.8 s) every minute during 3 days over a meadow. Receiving hydrophones recorded vocalizations from unknown fish inhabitants, which sound like /kwa/. Here, we address the impact of our measurements on this fish population. The authors' ears analyzed the recordings to identify all fish vocalizations and attempt to recognize /kwa/ individuals based on timbre. A comparative analysis of the /kwa/ sound production within time intervals between and during chirp transmission was made to understand if and how the sonic behavior is affected. As in few similar studies for other anthropogenic noise sources on a limited number of species, there was an observable effect. The major part of the vocalization patterns that were identified within the chirp time interval started near the onset of the frequency sweep, which first spans the range of /kwa/ sound production (0.4-2 kHz). The number of single vocalizations per unit time during that period was close to that during the no-chirp time interval while during higher swept frequencies was markedly smaller. This suggests the fish detection range may extend toward ultrasound to avoid dolphin predators. Quantitative results will be presented.

Session 4aBA

Biomedical Acoustics: Acoustic Imaging of Small Vessels and Low Speed Flow

Mahdi Bayat, Chair
Biomedical and physiology, Mayo Clinic, 200 1st St. SW, Rochester, MN 55905

Chair's Introduction—8:00

Invited Papers

8:05

4aBA1. Ultrasonic non-contrast perfusion imaging with adaptive demodulation. Jaime Tierney and Brett Byram (Biomedical Eng., Vanderbilt Univ., 2301 Vanderbilt Pl.; PMB 351631, Nashville, TN 37235, jaime.e.tierney@vanderbilt.edu)

Ultrasonic non-contrast perfusion imaging remains challenging due to spectral broadening of the tissue clutter signal caused by patient and sonographer hand motion. Simply, the velocity of the slowest moving blood has similar spectral support to the moving tissue. To address this problem, we developed an adaptive demodulation (AD) scheme to suppress the bandwidth of tissue prior to high-pass filtering. The method works by directly estimating the modulation imposed by tissue motion, and then removing that motion from the signal. Our initial implementation used single plane wave power Doppler imaging sequence combined with a conventional high-pass IIR tissue filter. However, other recent advancements in beamforming and tissue filtering have been proposed for improved slow-flow imaging, including coherent flow power Doppler (CFPD) and singular value decomposition (SVD). Here, we aim to evaluate AD separately as well as in comparison and in conjunction with improvements in beamforming and filtering using simulations and an *in vivo* muscle contraction experiment. We show that simulated blood-to-background SNR is highest when using AD + CFPD and a 100 ms ensemble, which resulted in a 9.88 dB increase in SNR compared to CFPD by itself. Additionally, AD + SVD resulted in a 54.6% increase in mean power within the *in vivo* muscle after contraction compared to a 6.84% increase with AD and a conventional IIR filter.

8:30

4aBA2. Enhanced sensitivity of flow detection in small vessels with slow flow by coherent-flow power Doppler. Jeremy J. Dahl (Radiology, Stanford Univ., 3155 Porter Dr, Palo Alto, CA 94304, jeremy.dahl@stanford.edu)

Power Doppler imaging is a commonly used method for the detection of flow. However, the sensitivity of power Doppler in small vessel detection is limited in clinical ultraound scanners by a combination of factors, including the small-diameter of vessels, slow flow through the vessels, the size of the Doppler ensemble, the presence of motion, the presence of noise, and the type of filter used. We have developed an alternative power Doppler imaging technique that enhances flow detection of small vessels and slow flow. This technique utilizes the same principles as power Doppler, but uses the coherence of backscattered blood signal rather than the backscattered signal power. We show that this technique, called coherent flow power Doppler (CFPD), is less sensitive to clutter-based signals and thermal noise and improves signal-to-noise ratio of the flow signal. We show through simulations, phantoms, and *in vivo* experimentation in human liver that this additional signal-to-noise ratio (7.5-12.5 dB) can be utilized to detect slower flow, improved frame rate (smaller Doppler ensemble), or improve the Doppler image profile. In addition, the technique demonstrates better signal-to-noise ratio for small-sized vessels (less than 2 mm).

8:55

4aBA3. Spatiotemporal clutter filtering methods for *in vivo* microvessel imaging. Pengfei Song and Shigao Chen (Dept. of Radiology, Mayo Clinic College of Medicine, Med. Sci. 1-26, 200 First St. SW, Rochester, MN 55905, song.pengfei@mayo.edu)

Ultrafast plane wave imaging enables acquisition and accumulation of a large number of Doppler ensembles in a short period of time, which substantially boosts ultrasound Doppler sensitivity by at least an order of magnitude. As a key component, spatiotemporal clutter filtering based on singular value decomposition (SVD) has recently demonstrated superior performance in tissue clutter suppression and microvessel signal extraction over conventional high-pass-filter-based clutter filtering techniques. This presentation introduces several novel SVD clutter filter techniques from our recent work, focusing on addressing the technical challenges (complex tissue motion, high computational cost, and noise) of SVD clutter filter during clinical translations of the ultrafast microvessel imaging technology for human imaging. We present a block-wise adaptive SVD filter method that can robustly threshold singular values on a local level, a real-time feasible SVD clutter filtering method based on partial SVD and randomized spatial downsampling, and a noise equalization and suppression method that can significantly improve microvessel imaging quality at a low cost. *In vivo* performance of these methods will be demonstrated in different organs.

9:20

4aBA4. New sampling and filtering approaches to label-free ultrasonic perfusion imaging. Yang Zhu (Dept. of BioEng., Univ. of Illinois at Urbana-Champaign, B412 Beckman Inst., 405 North Mathews Ave., Urbana, IL 61801, yangzhu2@illinois.edu), Minwoo Kim (Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), and Michael F. Insana (Dept. of BioEng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Color-flow and power Doppler imaging are safe, low-cost, non-invasive techniques for assessing blood flow throughout the body. Imaging microvascular flows and perfusion generally requires injectable contrast agents to separate tissue clutter from blood echoes. Our innovation is to modify echo acquisition to expand data dimensionality and increase sensitivity of non-enhanced PD imaging to very slow, disorganized flows. To separate perfusion from other echo signals, we developed a higher-order singular value decomposition filter that appropriately reduces data dimensionality. Effective elimination of clutter and noise results improved perfusion specificity. Our methods are applied to existing commercial US instruments, which makes them immediately valuable to medical practice. Our methods were previously tested at 24 MHz in a study involving muscle ischemia and highly-vascular melanoma lesions in mice. This study translates the methods to lower frequencies (5-10 MHz) for human trials. We developed a dialysis-cartridge device using TM-blood to quantifying very slow flow in the presence of tissue-like clutter and noise. We calibrated our technique to quantify the sensitivity, dynamic range, and spatial resolution of PD imaging. Results indicate an ability to separate flow speeds as low as 0.3 mm/s from clutter signals.

9:45

4aBA5. Multi-resolution rank analysis for non-contrast ultrasound imaging of slow flow. Mahdi Bayat (Biomedical and Physiol., Mayo Clinic, 200 1st St. SW, Rochester, MN 55905, bayat.mahdi@mayo.edu), Azra Alizad (Radiology, Mayo Clinic, Rochester, MN), and Mostafa Fatemi (Biomedical and Physiol., Mayo Clinic, Rochester, MN)

While conventional filtering techniques can effectively exclude tissue-related Doppler shifts from those resulting from blood motions in large vessels (flow rates ~ cm/s), significant spectral overlap can hinder proper clutter filtering in imaging slow flows (flow rates ~ mm/s or tenths of mm/s). Recently, several groups have proposed clutter filtering via processing of echo ensembles in the rank space instead of frequency domain. Though significant Doppler content can result from fast moving tissues, global coherence of the induced variations enables a low rank representation. In this talk, we first provide a brief overview of the low rank clutter removal methods. We then present a new framework for the analysis of Doppler ensembles using multi-linear decomposition of a tensor comprising Doppler ensembles at different rates. The model results in a high dimensional tensor from which, after proper rank selection, images of flow at different speeds can be extracted along different fibers. Using simulation models, we demonstrate the superior performance of the proposed method over singular value clutter filtering in imaging small vessels embedded in moving tissue undergoing both translational and shearing motions. We also present *in vivo* examples where the proposed method provides significant blood to clutter+noise ratio enhancement compared to singular value clutter filtering.

10:10–10:25 Break

Contributed Papers

10:25

4aBA6. Phantom study of flow speed measurement using ultrafast ultrasound plane wave imaging. Boran Zhou and Xiaoming Zhang (Radiology, Mayo Clinic, 321 3rd Ave SW, Rochester, MN 55902, Zhou.Boran@mayo.edu)

Color flow imaging is an ultrasound imaging technique widely used to image the blood movement through arteries and veins. Quantification of velocity is urgently demanding. The objective of this study is to use ultrafast plane wave imaging to measure the flow in speed *in vitro*. For the flow phantom, the diagonal vessel (2-mm inner diameter at 40° angle) was studied with injection of blood mimic fluid via syringe pump at various volume rates (15, 20, 25, and 30 ml/min). Verasonics Vantage system (Verasonics Inc., Kirkland, WA, USA) and a linear array transducer L11-4v (Verasonics Inc., Kirkland, WA, USA) were used for ultrasound acquisition, a 8-angle ($-7°$ to $7°$) plane wave compounding with effective pulse-repetition frequency (PRF) of 1000 Hz was used (central frequency = 6.42 MHz, number of transmit cycles = 4). A total of 600 ensembles were acquired. The frame rate was 1000/s.[XZ1] [BZ2] The ultrasound in-phase quadrature data size was $195 \times 156 \times 600$ (axial x lateral x temporal) with data type of complex double. Singular value decomposition (SVD)-based ultrasound blood flow clutter filters was used for ultrafast plane wave imaging.[XZ3] [BZ4] Randomized SVD was utilized to filter noise and tissue motion. Power Doppler image was generated. The color flow images were generated by using the 2-D autocorrelation method. Results showed that color flow measurement can quantitatively evaluate the flow speed in the tube of the flow phantom.

10:40

4aBA7. Quantitative morphological analysis of microvasculature in thyroid nodules using non-contrast high-resolution Doppler imaging. Siavash Ghavami (Radiology, Mayo Clinic, College of Medicine, 200 First St. SW, Rochester, MN 55905, roudsari.seyed@mayo.edu), Mahdi Bayat, Adriana Gregory, Bae Hyeong Kim, Kumar Viksit (Physiol. and Biomedical Eng., Mayo Clinic, College of Medicine, Rochester, MN), Jeremy Webb (Radiology, Mayo Clinic, College of Medicine, Rochester, MN), Mostafa Fatemi (Physiol. and Biomedical Eng., Mayo Clinic, College of Medicine, Rochester, MN), and Azra Alizad (Radiology, Mayo Clinic, College of Medicine, Rochester, MN)

We hypothesize that pattern of microvasculature in the thyroid nodules can be a biomarker of thyroid cancer. In this study, morphological features of microvasculature networks are assessed as potential biomarkers for differentiation of benign, suspicious and malignant thyroid nodules. This prospective study included n=81 patients, with a total of 61 benign, 12 suspicious and 8 malignant nodules verfied by pathology and cytology findings. Thyroid nodules were first identified using conventional B-mode ultrasound. A sequence of raw ultrasound data were acquired from each nodule site in two perpendicular planes at ~600 frames/sec. Power Doppler image of the microvasculature was formed using singular value decomposition clutter removal filtering. Gross images were further processed using a combination of morphology and Hessian-based vessel enhancement filtering. Background-free images of the microvasculature networks were segmented (vessels segment length $>385\mu$m and diameter $>90\mu$m) for morphological analysis. Using a Bayesian framework, measures of vessels density,

diameter and tortuosity were tested for significant differences in the three groups. Among all measures, the probability of vessel diameter $>600\mu$m in benign cases was significantly higher than that in malignant cases with the p-value <0.01 using Wilcoxon-Rank-Sum test, suggesting reliable cancer detection performance of the proposed method. [Acknowledgment: NIH Grant EB17213.]

10:55

4aBA8. Compound speckle model demonstrates two-phase wash-in of contrast-enhanced ultrasound cine loops. Matthew R. Lowerison (Medical Biophys., Univ. of Western ON, 95 Salem Pl., London, ON N6K 1T8, Canada, mloweri@uwo.ca), Ann F. Chambers (Oncology, Univ. of Western ON, London, ON, Canada), Hon S. Leong (Urology, Mayo Clinic, Rochester, MN), Nicholas E. Power (Surgery, Univ. of Western ON, London, ON, Canada), and James C. Lacefield (Robarts Res. Inst., London, ON, Canada)

During contrast-enhanced ultrasound (CEUS) imaging of tumor microvascular perfusion, nearby arteriole enhancement can be a dominant feature of the wash-in kinetics that obscures the perfusion characteristics of the capillary bed. This presentation demonstrates that statistical wash-in curves generated using our compound CEUS speckle model[1] exhibit two distinct phases corresponding to "fast-flow" and "slow-flow" enhancement that can be detected by fitting a linear combination of two monoexponential functions with different time constants. CEUS cine loops were acquired from a patient-derived xenograft model of renal cell carcinoma, where fresh tumor fragments were engrafted into the chicken embryo chorioallantoic membrane. Subharmonic CEUS images were acquired at 18 MHz using a destruction-replenishment protocol. Enhancement of manually segmented tumor cross-sections was analyzed offline in MATLAB. The CEUS cine loops from this xenograft model frequently exhibited in-plane arteriole enhancement. The two-phase fit discriminated a slow-flow component in 26.6% of time-intensity wash-in curves and 62.9% of the statistical wash-in curves, leading to a decrease in estimated tumor blood volume by 22.1% and 24.7%, respectively, compared to a simple monoexponential fit. These results suggest that conventional CEUS processing may frequently overestimate capillary blood volume. [1]M.R. Lowerison *et al*., Med. Phys. **44,** 99-111 (2017).

11:10–11:55 Panel Discussion

THURSDAY MORNING, 10 MAY 2018 — GREENWAY D, 10:10 A.M. TO 12:00 NOON

Session 4aEA

Engineering Acoustics, Signal Processing in Acoustics, and Underwater Acoustics: Advanced Transduction Technologies for Sonar Applications

Dehua Huang, Chair
NAVSEANPT, Howell St., Newport, RI 02841

Invited Papers

10:10

4aEA1. Transduction technologies for structural acoustics sonars. James Tressler (Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC 20375, james.tressler@nrl.navy.mil)

This presentation will describe underwater acoustic transduction technologies that the Naval Research Laboratory (NRL) has developed for structural acoustics sonars. The NRL "Skyfish" downward-looking sonar system is deployed on a 21-inch diameter autonomous underwater vehicle (AUV) that features a pair of extendable/retractable wings, each of which houses a thirty-two element cylindrical hydrophone array. The well-behaved acoustic source consists of a pair of transducers, each covering a separate frequency band. The source is designed to provide a fairly wide acoustic aperture for looking both fore and aft as well as in the port and starboard directions. Architectural details of the source and receiver, materials considerations, amplifier design, and experimental data will be presented. [This work supported by the Office of Naval Research & ESTCP.]

10:30

4aEA2. Design issues for eigenbeamforming spherical microphone arrays. Gary W. Elko (mh Acoust., 25A Summit Ave., Summit, NJ 07901, gwe@mhacoustics.com) and Jens Meyer (mh Acoust., Fairfax, VT)

Approximating an analytic function by orthonormal bases functions has its origins in the theory of least squares developed by Gauss and Legendre. The use of orthonormal bases functions has been a standard tool in soundfield analysis since Rayleigh. One common set of orthonormal bases functions that naturally fit the solution of the spherical wave equation are the spherical harmonics. In this talk we describe the concept of building a microphone array on a rigid spherical baffle. We process the output signals from the spherical array such that we spatially decompose the acoustic soundfield into spherical harmonic Eigenbeam signals. Computationally efficient Eigenbeamforming, or modal beamforming, with these processed Eigenbeam signals will be discussed. Two main design issues are the growth in sensitivity to self-noise for higher-order spherical modes at lower values of k*r where, k is the wavenumber and r is the radius, and spatial aliasing due to the finite number of microphones covering the spherical surface at higher values of kr. We will discuss how to practically handle these design issues and how these design solutions impact the overall performance of the Eigenbeams and the use of these components in the design of modal beamformers.

10:50

4aEA3. The design of broadband constant beamwidth and constant beam pattern transducers. Dehua Huang (NAVSEANPT, Howell St., Newport, RI 02841, DHHuang@cox.net)

The broadband constant beamwidth (CBT) and the constant beam pattern (CBP) transducers attracted much attentions, because of their main lobes or even the whole beam patterns (CBP case) are independent of frequencies. For a spherical (or hemispherical) design, the far field angular beam patterns are the same as the normal directional radial particle velocity, or shading, distribution on the surface of the spherical transducer, if the *ka* product for the wave number and the sphere radius is greater than one, per spherical Hankel function asymptotic approximation to the solution of the Helmholtz wave equation. Any arbitrary velocity shading functions can be expanded by series of Legendre polynomials for optimization beamwidth and sidelobe levels. This paper introduced a new transducer or array design, where the methodology by CBP is employed, while the geometry of CBT spherical cap is applied, such that the shading functions are no longer confined by only one Legendre polynomial $P(\cos(\theta))$ of the CBT case. Here, the shading examples by various Legendre polynomials of $P(z_0*\cos(\theta))$ and the classic Dolph-Chebyshev $T(z_1*\cos(\theta))$ of different orders have been successfully simulated, where the z_0 and the z_1 are the design control parameters of the beams.

11:10

4aEA4. A review of single crystal underwater transducers. Harold Robinson (NUWC Div. Newport, 1176 Howell St., Newport, RI 02841, harold.c.robinson@navy.mil)

The unique material properties of lead magnesium niobate-lead titanate (PMN-PT) single crystals enable compact, broadband, high power sound projectors that cannot be realized using conventional ceramics. This paper shall highlight single crystal transducer prototypes developed at the Naval Undersea Warfare Center, Division Newport, and elsewhere demonstrating that the broadband, high power performance promised by single crystal's material properties can be realized in practical underwater transducers. Various single crystal transducer designs for underwater applications, including longitudinal vibrators, fully active and active-passive air-backed cylinder transducers, free-flooded ring transducers and bender bar transducers, will be shown. These transducer designs cover a wide span of frequencies, and utilize both 33- and 32- operating modes. Tuned bandwidths of up to 2.5 octaves of frequency, as well as source level increases of up to 15 dB at the band edges, will be demonstrated. t will also be shown that these technologies provide stable performance under high drive and duty cycle conditions. Finally, the impact of using these single crystal sources on the drive electronics and power consumption shall also be discussed.

Contributed Papers

11:30

4aEA5. Optical breakdown as a broadband underwater acoustic source. Athanasios G. Athanassiadis (Dept. of Mech. Eng., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Bldg. 3-257c, Cambridge, MA 02139, thanasi@mit.edu)

Current underwater transducers provide little support for remote measurement of material properties. Such measurements are often limited by high impedance contrast, and target inhomogeneities that arise from biofouling and corrosion. In the presence of these limitations, a sensing system would benefit from broadband, spatially-localized target excitation. The high bandwidth would allow a return signal to encode broader information content about the target, while spatial localization would provide stronger target excitation and higher spatial resolution for measurements. Although these source properties are not achievable with traditional transducer designs, they can be produced through laser-induced optical breakdown in water. During optical breakdown, focused laser light excites a localized plasma, which expands and produces a mechanical shock wave. The shock properties are set by both the medium hydrodynamics and the laser parameters, allowing the acoustic source to be optically tuned. In laboratory experiments, breakdown from a 4 mJ/ 4ns laser pulse emits an exponential pressure wave with a decay constant under 250 ns and peak pressure near 600 kPa at 2 cm range. Here, I discuss the physics and fundamental limitations of acoustic transduction via underwater optical breakdown. Experimental results illustrate how the acoustic source properties can be tuned through optical control.

11:45

4aEA6. High power evaluation of textured piezoelectric ceramics for SONAR projectors. Arjun Shankar (Penn State Univ. / McKay Conant Hoover, Inc., 242 S Beck Ave, Tempe, AZ 85281, arjunkshankar@gmail.com) and Richard Meyer (Penn State Univ., State College, PA)

Textured Ceramics of the relaxor ferroelectric PMNT have shown exceptional piezoelectric properties suitable for high power SONAR projectors. The piezoelectric coefficient (d33>800pm/V), the electromechanical coupling coefficient (k33>0.80), and the mechanical quality factor (Qm>90) in various forms of PMNT showcase the potential for these textured ceramics to be implemented in a high power, broad bandwidth acoustic projector. This work attempted to devise a set of experiments to quickly characterize the electromechanical performance of these materials in relevant devices in order to provide feedback to both the material developers and the user community. In this effort, textured PMNT and Manganese doped PMNT (Mn:PMNT) were compared to the performance of conventional lead zirconate titanate (PZT) ceramics. Linearity and harmonic distortion measurements were made in a high pressure test facility to measure source levels of these SONAR projectors. In addition, isothermal measurements were made at 50 degrees Celsius to monitor source levels as a function of duty cycle. The results from these measurements were then compared to modeled results and appropriate recommendations on material composition improvements were made.

Session 4aID

Interdisciplinary, ASA Committee on Standards, Noise, Structural Acoustics and Vibration, and Architectural Acoustics: Acoustical Standards in Action: Realization, Applications, and Evolution

Christopher J. Struck, Chair
CJS Labs, 57 States Street, San Francisco, CA 94114

Invited Paper

8:00

4aID1. An overview of the Standards Program of the Acoustical Society of America. Christopher J. Struck (CJS Labs, 57 States St., San Francisco, CA 94114, cjs@cjs-labs.com)

An overview of the standards program of the Acoustical Society of America is presented. The American National Standards Institute accreditation and the U.S. voluntary consensus process are described. The structure of the ASA standards organization is explained and the roles of each standards committee are outlined. The standards development process from new work item proposal through final approval is detailed. The relationship between national and international standards bodies is also discussed. Examples of the practical benefits of acoustical standards for both participating organizations and end users are given. Information about how to participate in the standards process is also provided.

Contributed Paper

8:20

4aID2. Laboratory accreditation—Building confidence on acoustical testing- and calibration-lab quality. Richard J. Peppin (Engineers for Change, Inc., 5012 Macon Rd., Rockville, MD 20852, PeppinR@asme.org), George Anastasopoulos, and Prasanth S. Ramakrishnan (Int. Accreditation Service, Inc., Brea, CA)

Testing (and calibration) laboratories, to be good, must provide consistent accurate and precise measurements of the test objects based on their well-defined procedures. Lab accreditation is one way to assure the engineers, scientists, and technicians in the lab AND the lab customers and the public have confidence in the skills, facilities, and results of the lab's work. There is an international standard, ISO/IEC 17025-2005 "General requirements for the competence of testing and calibration laboratories." This standard is for use by laboratories in developing their management system, for quality, administrative, and technical operations Following this standard will help labs assure quality in their work. To assure the public (and the lab's customers) that the lab does follow this, an objective agency evaluates the lab's quality program. This process is called "accreditation." Accreditation is done under strict quality control by accrediting agencies, of which there are several available for worldwide labs. Most of the presentation will emphasize what is required and involved in the accreditation process This presentation will also discuss (1) why accreditation is useful, (2) who does the accreditation, (3) how are they qualified to do accreditation, (4) how costs are determined, and (5) how fast can it be accomplished.

Invited Paper

8:35

4aID3. Acoustical standards in the national parks. Kurt M. Fristrup (Natural Sounds and Night Skies Div., National Park Service, 1201 Oakridge Dr., Ste. 100, Fort Collins, CA 80525, kurt_fristrup@nps.gov)

U. S. National Parks began actively conserving park acoustic resources decades after U. S. community noise management practices were established. This presented opportunities to benefit from the knowledge acquired in community noise settings and challenges to adapt this knowledge and devise new methods as appropriate for national park settings. Participation in ANSI/ASA accredited standards development processes provided numerous benefits to the National Park Service (NPS): Access to a substantial fraction of the world's experts in acoustics, noise measurement, and noise reduction, many of whom were deeply involved in the development of community noise standards, Participation in an open, formal, coordinated, consensus-based process to develop voluntary standards, Engagement of a diverse range of stakeholders to develop standards through processes of cooperation and compromise, Conformance with the Office of Management and Budget's requirement to use voluntary consensus standards in lieu of government-unique standards (Circular A-119). NPS is an institutional member of two accredited standards committees and participates in the leadership of one committee and one working group. One standard has been published from this working group (ANSI S3-SC1.100 and ANSI S12.100: Methods to Define and Measure the Residual Sound in Protected Natural and Quiet Residential Areas). A second standard is in development.

Contributed Papers

8:55

4aID4. Comparing the similarity of sharpness predictions based on specific loudness from ANSI S3.4-2007 with those according to DIN 45692:2009 for several real-world sound classes. S. Hales Swift and Kent L. Gee (Phys. and Astronomy, Brigham Young Univ., N221 ESC, Provo, UT 84602, hales.swift@gmail.com)

A previous paper [Swift and Gee, Proc. Mtgs. Acoust., p. 030001, 2017] showed methods for transforming the standardized form of sharpness to accept inputs from the ANSI S3.4-2007 loudness standard. A further recent publication [Swift and Gee, JASA-EL, 2017] showed additional validation results comparing the results of the simplest version of the transformed metric for a set of 40 real-world sounds. In this paper, three versions of the ANSI-based sharpness metric are analyzed, in order to determine which of the three implementations gives the most similar results to the original DIN 45692:2009 standard. Finally, classes of sounds are identified for which predictions between the three ANSI-based sharpness metrics and DIN 45692:2009 may differ to a greater or lesser extent.

9:10

4aID5. Improving practices through analysis of long-term microphone calibration records. Gordon M. Ochi, Matthew G. Blevins, Gregory W. Lyons, and Edward T. Nykaza (Construction Eng. Res. Lab., U.S. Army Engineer Res. and Development Ctr., 2902 Newmark Dr., Champaign, IL 61822, Gordon.M.Ochi@erdc.dren.mil)

A microphone's history of calibration is a useful source of information, as an independent analysis of the data can reveal more than simply the metrological traceability. The U.S. Army Engineer Research and Development Center maintains over 13 years of calibration records from an inventory of over 280 Class 1 microphones, which were examined for historical validity and consistency. Analyzing trends in the laboratory reported sensitivities showed that 12.9% of active microphones exhibited four possible trends that could indicate potential microphone failure: sensitivity drift, sensitivity drift followed by instability, jump in sensitivity, and general instability. Many of the microphones that displayed these trends had uncertainties greater than a ± 1.0 dB tolerance inferred from ANSI S1.14 Class 1 microphone specifications, and an additional 10.7% of microphones showed errors in the calibration laboratory's transcription process or testing procedures. Recommendations are put forth for best practices relative to laboratory independent microphone calibration record documentation and analysis, and a quantitative procedure for flagging microphones was developed. Specifically, it is recommended that microphones with a component type 'A' uncertainty due to the history of calibrations that exceeds ± 1.0 dB be removed for further inspection, in reference to the Class 1 microphone specifications in ANSI S1.14.

Invited Papers

9:25

4aID6. Pressure reciprocity calibrations of laboratory standard microphones at the National Institute of Standards and Technology. Randall Wagner (National Inst. of Standards and Technol., 100 Bureau Dr, MS 6833, Gaithersburg, MD 20899, randall.wagner@nist.gov)

The reciprocity technique has long served as a method for pressure calibration of microphones. It is a primary method, which determines microphone sensitivities from first principles and does not require a previously calibrated acoustic transfer standard. For calibrations of laboratory standard microphones, this method is standardized and utilized at national measurement institutes worldwide. Standard microphones calibrated by reciprocity are in turn used to calibrate additional microphones and sound calibrators, which apply known sound pressures to calibrate acoustical measuring devices and systems. Reciprocity calibrations done at the National Institute of Standards and Technology (NIST), which is the national measurement institute for the U.S., provide its customers with accurate results traceable to the International System of Units (SI). These customers and organizations that utilize their services perform large numbers of secondary and further calibrations and measurements concerned with hearing conservation and testing, aircraft noise, noise regulation enforcement, acoustical test and measurement equipment, and auditory research. An overview of the reciprocity technique is presented along with a few examples of how customers of the NIST acoustical calibration services make use of their calibrated devices and calibration data.

9:45

4aID7. Types of regulatory tools and their advantages, limitations, and disadvantages in a Model Noise Ordinance Standard. Leslie D. Blomberg (Noise Pollution Clearinghouse, Box 1137, Montpelier, VT 05601-1147, les@nonoise.org)

The American National Standards Institute is developing a model community noise ordinance. There are seven types of regulatory tools available to communities to regulate noise that can be used in a model noise ordinance. These include (1) nuisance based regulations, (2) disturbing the peace regulations, (3) decibel based regulations, (4) plainly audible regulations, (5) operational restrictions and prohibitions, (6) equipment requirements, and (7) quiet zones. In this paper, each is described, as well as the advantages, limitations, and disadvantages of each tool. A *Matrix of Advantages of Regulatory Tools Used in Community Noise Regulation* is presented and provides easy comparison between the regulatory tools. This matrix can provide the basis for a proposed Model Noise Ordinance standard.

10:05–10:20 Break

10:20

4aID8. Testing the limits of reverberation room qualification standard AHRI 220. Derrick P. Knight (Ingersoll Rand - Trane, 824 Olympic Dr., Onalaska, WI 54650, derrick.knight@irco.com)

Trane's acoustic lab recently upgraded measurement equipment which necessitated re-qualification of two reverberation chambers according to AHRI 220. We took this opportunity to test restrictions listed in the standard in relation to source-to-wall spacing and minimum number of fixed microphone positions. Measurements were simultaneously taken using rotating and fixed microphones. It will be shown that the wall proximity limit is overly restrictive, while the minimum fixed microphone requirement is appropriate.

10:40

4aID9. SII—Speech intelligibility index standard: ANSI S3.5 1997. Caslav Pavlovic (BatAndCat Corp., 602 Hawthorne Ave., Palo Alto, CA 94301, chas@batandcat.com)

The Speech Intelligibility Index Standard (SII) defines a method for computing a physical measure that is highly correlated with the intelligibility of speech. The SII is calculated from acoustical measurements of speech and noise. In this talk the activities of the Working Group for the Speech Intelligibility Standard will be reviewed including the issues to be resolved, the changes, and the additions under consideration. The general framework of the current standard (ANSI S3.5-1997) is such that new methodologies for specifying the effective ("equivalent") speech and noise can be easily incorporated as long as they have been tested and found valid for a well specified set of circumstances. The history of the Speech Intelligibility Index (SII) Standard and its relationship to the Articulation Index (AI) and the Speech Transmission Index(STI) will also reviewed. The general philosophy of Speech Intelligibility Index will also be discussed including the definition and specification of its fundamental variables: dynamic speech and noise spectra, information transmission by different bands, and their biologic/evolutionary compatibility with both the threshold of hearing, and the neural tuning.

11:00

4aID10. History of the sound level meter in standards. Leslie D. Blomberg (Noise Pollution Clearinghouse, Box 1137, Montpelier, VT 05601-1147, les@nonoise.org)

In 1932, the Acoustical Society of America started work on the first American Standards Association standard for sound level meters. The purpose was to create a set of standards for sound level meters "such that, if a given noise of a general character is measured with any meter designed in accordance with the standards, the result will be substantially the same as that which would be obtained with any other similarly designed meter." The standard that resulted, Z 24.3 1936 was the first of many that trace the development of the sound level meter over the past 86 years. This paper follows the history and development of the sound level meter in the original American Standards Association and subsequent American National Standards Institute standards: Z 24.3 1944, S1.4 1961, S1.4 1971, 1.4 1983, and S1.4 2014. During that time the sound level meter has shrunk from a more than 100 pound, vacuum tube machine to a modern the modern, lightweight meter. The standard, however, has grown from a mere 11 pages in 1936 to 50 pages in 2014.

Session 4aMU

Musical Acoustics: General Topics in Musical Acoustics

Taffeta M. Elliott, Chair
Psychology Dept., New Mexico Institute of Mining and Technology, 801 Leroy Place, Socorro, NM 87801

Contributed Papers

8:30

4aMU1. Art in Acoustics. Peter Francis C. Gossweiler (Instituto de Artes, UFRGS, Trav. Nsa. Sra. de Lourdes, 230 apt. 503 D, Porto Alegre, Rio Grande do Sul 91920-040, Brazil, petergossweiler@gmail.com)

Since the first time the caveman shouted out loud to the heart of a cave, to listen for a kind of magical relationship, a huge spectrum of possibilities has grown to deal with this phenomenon of nature. In fact, the eco or even the reverberation has been fascinating the human being till nowadays, as we can see on noise control studies, acoustics studies, sound studies, etc. But we propose another approach, maybe not so straight related to science (as a demand centered on results) but with an undeniable interest with its capacities of creation through the study of the acoustics (with a demand focused on the process). From artistic works of John Cage, Alvin Lucier, Nicolas Collins, Charlemagne Palestine, Llorenç Barber, and Lukas Kühne, we engage an interdiciplinary relation between art and acoustic to clarify this type of magic in our life. A historical, theorical, and critical analysis is applied in this task.

8:45

4aMU2. Examining the effect of emerging audio formats and technology on acoustical space in film composition and mixing. Conor P. Cook (Design & Eng., Acoust. Surfaces, 123 Columbia Court N., Chaska, MN 55318, ccook@acousticalsurfaces.com)

Because film audio must balance the three basic soundtrack elements of dialogue, music, and effects, acoustical masking and professional disagreements during the post-production mixing process affect the importance given to the film score. Dialogue is customarily and logically given priority, causing film composers and music mixers to work around the acoustical space taken up by speech, typically through panning, equalization, avoidance of speech frequencies, and "density" in orchestration. Improvements in audio systems and formats, such as surround sound and now Dolby Atmos, are affording composers and mixers more acoustical space to avoid masking effects without a loss of dramatic support for the film, including the introduction of height cues and greater surround panning control, as well as sound objects, which control the localization of discrete sounds in 3D space. The new technologies also present acoustical design challenges as they interact in surprising ways with specific "acoustic signatures" of widely-varying media presentation spaces. Acoustical environment examples from earlier and recent film mixes as well as critical literature are given. Composers and mixers are encouraged to explore the range of expressive acoustical options made available by emerging technologies and the more widely-compatible mixing spaces designed for them.

9:00

4aMU3. Laryngeal height, modal registers, and modes of phonation contribute jointly to vocal timbre. Taffeta M. Elliott (Psych. Dept., New Mexico Inst. of Mining and Technol., 801 Leroy Pl., Socorro, NM 87801, taffeta.elliott@nmt.edu) and Lisa Popeil (Voiceworks, Sherman Oaks, CA)

Timbre encompasses every distinguishing quality of a sound other than its pitch, duration, loudness, and sound location. Musical sound makers exploit contrasts in timbre. The voice organ in particular is capable of an impressive array of vibrational patterns with acoustic consequences perceived as timbral changes. As a first step towards determining the dimensionality of perceptual timbre in a singing human voice, we assessed the relation between, on the one hand, production parameters, executed by a well-trained vocalist capable of the articulation control necessary for demonstrations, and on the other hand, the complex spectral and temporal modulations in the acoustic signal of the resulting 50 timbral variants. Production parameters on a high front vowel included 5 laryngeal heights (from the lowest possible to a mostly elevated position), 4 modes of phonation (clean, blowy, breathy, and pressed), and several changes to resonance (nasal, bright, and muted) as well as vocal register differences comparing vocal fry and modal registers. Raised and lowered larynx positions alter formants in a regular manner that overlaps with phonation patterns and other articulatory alterations of resonance.

9:15

4aMU4. Articulatory correlates of the acoustic transition during the second passaggio of sopranos. Richard C. Lissemore, Kevin Roon (Speech-Language-Hearing Sci., Graduate Center/The City Univ. of New York, 499 Fort Washington Ave., Apt. #4F, New York, NY 10033, rlissemore@gradcenter.cuny.edu), Christine H. Shadle (Haskins Labs., New Haven, CT), and D. H. Whalen (Speech-Language-Hearing Sci., Graduate Center/The City Univ. of New York, New Haven, Connecticut)

The present experiment examines articulatory correlates associated with the presence or absence of a change in A_1-A_2, the amplitude difference between the first two harmonics (f_o and $2f_o$),in the second passaggio transition (D5♮ to F5♮, or 587 to 698 Hz) of the soprano voice. An earlier investigation of this area in a classical, techniqued soprano singing on a low vowel revealed a change from negative to positive values in A_1-A_2 at a pivot point between E5♭ and E5♮. But when the same singer produced sounds without technique, the A_1-A_2 values remained negative. The present articulatory investigation uses ultrasound and optical tracking for analysis of head-corrected tongue contours, jaw opening, lip aperture, and lip protrusion that accompany the presence or absence of the acoustic change in this same singer. Preliminary results replicate the relationship between the acoustic change from negative to positive values of A_1-A_2 and show a substantial change in tongue dorsum height. In contrast, singing that exhibits only negative values of A_1-A_2 shows no such articulatory changes. The same experimental protocol was then used with sopranos who can and cannot make the A_1-A_2 acoustic change. Acoustic and articulatory results will be reported. [Funded by NIH grant DC-002717.]

4a THU. AM

9:30

4aMU5. Comparing clarinet grunt and squeak notes to transitions in coupled oscillators. Cherise Cantrell, Joshua Vawdrey (Phys., Utah Valley Univ., MS 179, 800 W University Pkwy, Orem, UT 84058, cantrell.cherise@gmail.com), Jeffrey O'Flynn (Music, Utah Valley Univ., Orem, UT), and Bonnie Andersen (Phys., Utah Valley Univ., Orem, UT)

Under certain circumstance, a clarinet can cause an undesired squeak or grunt note. A squeak note is the clarinet shifting to a higher register—a higher set of harmonics—while a grunt note is the clarinet shifting to a lower register. One might hypothesize that such changes occur naturally in the clarinet over time. Analysis of such transitions with waterfall plots reveals a similarity to transitions made with thermoacoustically-driven coupled oscillators. For coupled oscillators, transitions can occur when the resonance of one oscillator passes through that of the other. This suggests that the cause of the undesired transitions in the clarinet are the result of one resonance (possibly the reed, which is affected by the embouchure) passing through that of the resonance of the other (likely the bore) rather than the clarinet itself relaxing to a different register as a function of time. Notes played on an artificially-blown clarinet in the second register were shown to be stable over several minutes. This suggests that changes in the embouchure drive the clarinet to grunt or squeak.

9:45–10:00 Break

10:00

4aMU6. Computational analysis of the Bundengan, an endangered musical instrument from Indonesia. Indraswari Kusumaningtyas (Dept. of Mech. and Industrial Eng., Universitas Gadjah Mada, Jalan Grafika 2, Kampus UGM, Yogyakarta, DIY 55281, Indonesia, i.kusumaningtyas@ugm.ac.id) and Gea O. Parikesit (Dept. of Nuclear Eng. and Eng. Phys., Universitas Gadjah Mada, Yogyakarta, DIY, Indonesia)

The Bundengan, an endangered Indonesian musical instrument, was first developed and played by duck herders. The dome-shaped resonator, which also protects the herders from adverse weather, is made from a specially woven grid of bamboo splits, covered with layers of bamboo sheaths, held in place with sugar palm fibre straps. Inside the resonator are a set of long, thin bamboo plates and a set of strings equipped with small bamboo clips, which vibrate together with the strings. When playing, the Bundengan player uses one hand to pluck the clipped strings and the other to pluck the bamboo plates. The clipped strings generate metal-like sounds, while the bamboo plates generate drum-like sounds. Hence, the Bundengan can imitate the sound of gamelan, a traditional Indonesian instrumental ensemble. Bundengan players report difficulties to tune this instrument and to keep it in tune for a long time. We develop simulations to help Bundengan musicians better understand this instrument. The first simulation quantitatively predicts how the bamboo clips influence the frequencies of the clipped strings. The second one quantitatively predicts how the shape of the resonator affects the soundscape of the Bundengan. These simulations also allow for a more precise procedure to build the instrument.

10:15

4aMU7. Switch of synchronization states of aeroacoustical coupled organ pipes. Jost L. Fischer and Rolf Bader (Inst. of Systematic Musicology, Univ. Hamburg, Neue Rabenstr. 13, Hamburg, Hamburg D-20354, Germany, jost.leonhardt.fischer@uni-hamburg.de)

The separation distance of two equally tuned, aeroacoustically coupled organ pipes is varied stepwise in a synchronization experiment. At distances of multiples of about half the wavelength of the fundamentals, abrupt changes of SPL and fundamental frequency of the coupled and synchronized system can be observed. Frequency jumps also occur at higher harmonics. The nonlinear coupling function of the system implicitly takes into account the time delay of the system caused by the separation distance and therefore plays an important role for the observed phenomenon. The effect can be explained as a switch of synchronization states from in-phase to anti-phase and vice versa.

10:30

4aMU8. A comparison of fractional-sized to full-sized cellos. Thomas Blanford (Graduate Program in Acoust., The Penn State Univ., 1623 S. Ashwicken Ct, State College, PA 16801, teb217@psu.edu) and Micah R. Shepherd (Appl. Res. Lab, Penn State Univ., State College, PA)

Fractional-sized cellos (3/4, 1/2, etc.) are designed for the same musical playing range as a full-sized cello (4/4) but with scaled proportions for players for whom a full sized cello is too large. To compensate for the shorter string length of the smaller instruments, the strings are adjusted in order to obtain the correct tuning. The cello body vibration, which is strongly coupled to the internal air cavity, would not be expected to scale in the same manner as the strings. This causes the bridge impedance seen by the strings on the fractional-sized cellos to differ from the bridge impedance seen by the strings on a full-sized cello. In this talk, the physical dimensions of a $^1/_2$ and $^3/_4$ cello are compared with a full cello. Drive point measurements are also compared to illustrate how the strings couple differently with body of each size cello. The fractional-sized cellos are found to exhibit a slightly different sound due to the bridge impedance mismatch.

10:45

4aMU9. Correlation of string/body resonances on a cello. Samantha A. Young (Loyola Univ. Chicago, 1209 W Arthur Ave., Apt 209, Chicago, IL 60626, syoung11@luc.edu)

This project focuses on a 2001 Garavaglia full size acoustic cello, and investigates the resonance properties of a wooden body. Previous studies have estimated that the body resonance lies between the range of D_3-$F_3^{\#}$. The goals are to investigate the radiation patterns of the produced sound waves, to take high-speed video of a played string to physically observe the standing wave, and to simulate the body resonance of the cello. A mathematical model is formed to approach the situation where a musician plucks a string. Typical approaches assume the surface area of a finger to be a point source, but the surface area is actually parabolic, forming a bend in the string that is continuous to the following straight line. High speed video was taken to try out methods for tracking a standing wave at 1000 frames per second for a maximum number of data points per period. We will correlate the waveform fits from the high speed videos to the Fourier transformations from the acoustic data. These same methods will be applied in an anechoic chamber to test "ideal" conditions in order to isolate the body resonances. We will correlate the data from body resonances to our string data.

11:00

4aMU10. Impact of internal damping on forced vibrations of piano soundboards. Rolf Bader (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, R_Bader@t-online.de), Niko Plath, and Florian Pfeifle (Inst. of Musicology, Univ. of Hamburg, Hamburg, Hamburg, Germany)

A Finite-Difference Model of a grand piano soundboard including internal damping is used to estimate the influence of the damping on the vibration of the soundboard. Theoretically, the increase of damping reduces eigenmodes and leads to forced oscillation patterns. Those patterns show considerable dependency of their vibrational shapes upon the driving point. In the extreme case of a heavy damping, the wave starting at the driving point does not reach the boundaries of the soundboard with much strength and is therefore not reflected. Then no eigenmodes exist anymore and the plate behaves like an infinite plate. In the other extreme of a soundboard not damped at all the eigenmodes of the soundboard clearly appear. The measured patterned of a soundboard are compared to the modeled ones and conclusions are derived in terms of the amount of damping in a real soundboard.

Session 4aNS

Noise, Psychological and Physiological Acoustics and ASA Committee on Standards: Hearing Protection: Impulse Peak Insertion Loss, Specialized Hearing Protection Devices, and Fit-Testing I

Cameron J. Fackler, Cochair
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Elliott H. Berger, 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—8:00

Invited Papers

8:05

4aNS1. Total hearing health for preventing occupational hearing loss. William J. Murphy (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov)

The National Institute for Occupational Safety and Health (NIOSH) developed the Total Worker Health program that encourages health and safety professionals to consider more than just a particular exposure. Workers can suffer hearing loss due to both workplace, but due to a wide range of activities outside of work. Training workers in the proper use and application hearing protection can extend the effectiveness of hearing loss prevention efforts and promote the concept of Total Hearing Health. Awareness of hazardous activities such as the use of power tools, firearms, and chemical exposures, can instill a safety culture beyond the workplace. Hearing protector fit testing gives workers an increased self-efficacy and assurance that they can safely enjoy noisy activities without the need to sacrifice their ability to hear. This presentation will describe the work that NIOSH has undertaken to promote hearing protector fit-testing for the workplace and describe ways in which the Centers for Disease Control and Prevention has promoted hearing health.

8:25

4aNS2. Fit-testing systems for hearing protectors: ANSI standards to the rescue. Cameron J. Fackler and Elliott H. Berger (3M Personal Safety Div., 7911 Zionsville Rd., Indianapolis, IN 46268, cameron.fackler@mmm.com)

To standardize the performance of fit-testing systems for hearing protection devices (HPDs), an ANSI/ASA standard has been in development since 2008. As of this writing, BSR/ASA S12.71 "Performance criteria for systems that estimate the attenuation of passive hearing protectors for individual users" has been submitted for public comment and ballot. This standard refers to fit-testing systems as field attenuation estimation systems (FAESs). In this talk, we introduce important FAES concepts, as handled in the draft S12.71. There are two types of FAESs: physical systems that take an objective measurement and psychophysical systems that rely on subjective responses from the user being tested. FAESs may produce a quantitative output as a personal attenuation rating (PAR) or simply indicate pass/fail for a given HPD fit. For all types of systems, S12.71 specifies methods to assess the quality of the attenuation estimates, by comparing the FAES results to a laboratory real-ear attenuation at threshold (REAT) measurement. Other important considerations covered in S12.71 include the maximum levels of ambient noise in which FAESs may be operated, the incorporation of uncertainties due to HPD fitting and spectral variability in users' noise exposures, calibration intervals, and information that must be provided to users.

8:45

4aNS3. A field microphone in real ear hearing protector attenuation measurement system. Kevin Michael (Michael & Assoc., Inc., 2766 W. College Ave., Ste. 1, State College, PA 16801, kevin@michaelassociates.com)

A Field-Microphone in Real Ear (F-MIRE) hearing protector Field Attenuation Estimation System (FAES) was developed for individual or group testing of any type or style of hearing protector device (HPD). Both occluded and unoccluded ear canal stimulus level measurements are made with a tiny Micro Electro-Mechanical System (MEMS) microphone mounted on a 2-mil thick flat flexible cable. The microphone is mounted inside of a conventional foam ear dam to protect the microphone and the ear canal and to facilitate insertion into the ear canal. The presence of the cable in the ear canal was demonstrated to have a negligible effect on attenuation measurements for both muff and insert-type HPDs. The system measures HPD attenuation in 1/1 or 1/3 octave bands and it is capable of simultaneously measuring HPD attenuation on up to eight human test subjects. Stimuli can be presented via headphones or loudspeakers. Human subject testing using both the laboratory based ANSI S12.6-2016 method and the F-MIRE system was performed using muff- and insert- type HPDs. Regression equations were generated from these data. These equations are incorporated in the F-MIRE software to allow valid comparisons between F-MIRE data and labeled HPD attenuation data.

9:05

4aNS4. Fit testing of a hearing protection device with integrated in-ear noise exposure monitoring. Christopher J. Smalt, Shakti K. Davis (MIT Lincoln Lab., 244 Wood St, Lexington, MA 02420, Christopher.Smalt@ll.mit.edu), William J. Murphy, Chucri A. Kardous (Hearing Loss Prevention Team, CDC-NIOSH, Cincinnati, OH), Joe Lacirignola, and Paul Calamia (MIT Lincoln Lab., Lexington, MA)

In-the-ear hearing protectors are often used in high-noise environments, such as in military operations and during weapons training. The fit of such a hearing protection device in the ear canal can cause significant variability in the noise dose experienced, an important factor in characterizing the auditory health risk. Our approach for improved dose estimates under these conditions involves a portable noise recorder used to capture in-the-ear noise behind a hearing protector, and on-body noise, while assessing hearing protection fit throughout recording. In this presentation we describe evaluation of this system using ANSI S12.42 testing using a shock tube and an acoustic test fixture, to evaluate impulse peak reduction and for measurement validation. Also explored were angle dependent effects on the peak insertion loss and measurement accuracy of the on-body recorder. An exploratory study was conducted with a small sample of experimenters during a recent Navy-sponsored noise survey conducted at Marine Corps Base Quantico. Results will be presented from a shooter's ear and a nominal instructor's position as well as from bystanders observing at a firing range. Our results show good correspondence between behavioral fit-testing of insertion loss and estimated protection from the in-ear and on-body microphones. [Work supported by ONR.]

9:25

4aNS5. Acoustical corrections to be used for improved in-ear noise dosimetry measurements. Fabien Bonnet (École de Technologie Supérieure, 1100 Rue Notre-Dame Ouest, Montréal, QC H3C 1K3, Canada, fabien.bonnet@etsmtl.ca), Hugues Nélisse (Institut de recherche Robert-Sauvé en santé et en sécurité du travail, Montreal, QC, Canada), and Jeremie Voix (École de Technologie Supérieure, Montréal, QC, Canada)

Although much effort has been made to reduce sound at the source, hearing protection devices (HPDs) remain currently the most widely used solution against occupational noise exposure in an attempt to prevent noise-induced hearing loss. Such devices, though, often suffer from uncertainties when it comes to ensuring ideal protection for a given individual. These uncertainties stem mainly from three unknown variables: i) the actual attenuation provided by the HPD to the individual; ii) the ambient noise levels; iii) the wear time of the given HPD. In-ear dosimetry (IED) could overcome these issues, as it makes it possible to establish precise personal noise exposures through real-time in-ear noise monitoring. Furthermore, IEDs can be worn in the open, occluded (under earplugs), or cushioned (under earmuffs) ear. Nevertheless, current IEDs are weakened by two inaccuracies: the non-discrimination of self-induced noise, and the lack of acoustical corrections to convert the measured in-ear sound pressure levels to the eardrum and to the free-field. The latter issue is treated in this presentation, which describes methods to identify the appropriate acoustical corrections to be used for improved IED. Such methods aim at personal calibration procedures, so that the unique ear geometry of each individual is accounted for. Preliminary results will be presented as a comparison between theoretical and experimentally measured corrections on human test-subjects.

9:45–10:00 Break

10:00

4aNS6. Earplug comfort: From subjective assessment on the field to objective measurement and simulation using augmented artificial heads. Olivier Doutres (Mech. Eng., École de technologie supérieure, 1100 rue Notre-Dame Ouest, Montréal, QC H3C 1K3, Canada, olivier.doutres@etsmtl.ca), Franck C. Sgard (IRSST, Montreal, QC, Canada), Simon Benacchio (Mech. Eng., École de technologie supérieure, Montréal, QC, Canada), Jonathan Terroir (INRS, Vandoeuvre Les Nancy, France), Nellie Perrin, Nicolas Trompette (INRS, Vandoeuvre-lès-Nancy, France), Alessia Negrini (IRSST, Montreal, QC, Canada), Marc-André Gaudreau (Mech. Eng., Université du Québec à Trois-Rivières, Montréal, QC, Canada), Caroline Jolly (IRSST, Montreal, QC, Canada), Alain Berry, Philippe-Aubert Gauthier (Mech. Eng., Univ. of Sherbrooke, Sherbrooke, QC, Canada), Thomas Padois (Mech. Eng., École de technologie supérieure, Montréal, QC, Canada), and Chantal Gauvin (IRSST, Montreal, QC, Canada)

For several years, lack of comfort has been pointed out as a major reason of earplug poor efficiency as noise control solution. The great complexity of comfort makes it difficult to predict it in the earplugs design phase. It is rather considered in an empirical way by the manufacturers using trial-and-errors approaches based on subjective assessment over a panel of subjects. Furthermore, because comfort is not quantified, Occupational Health and Safety (OHS) practitioners cannot select earplugs ensuring wearer comfort. To address these issues, a major international research project funded by two OHS institutes (IRSST in Canada and INRS in France) and involving several Universities (in Canada and England) started in 2017. The main objectives are to: (1) improve the understanding of earplugs comfort as perceived by field workers with consideration of all comfort components, (2) develop laboratory tools (augmented experimental and virtual artificial heads) to measure physical design variables related to the auditory, physiological, and functional components of comfort and (3) design a battery of hybrid objective/subjective comfort indices to quantify / measure / predict the different components of comfort. The aim of this presentation will be to present the project and first results.

10:20

4aNS7. Application of a registration method on magnetic resonance images to evaluate the displacement field of a human subject ear canal due to various earplug insertions. Simon Benacchio, Olivier Doutres (École de technologie supérieure, 505 Boulevard de Maisonneuve O, Montréal, QC H3A 3C2, Canada, Simon.Benacchio@irsst.qc.ca), Arthur Varoquaux (Aix Marseille Université, CNRS, CRMBM/CEMEREM UMR 7339, Marseille, France), Eric Wagnac (École de technologie supérieure, Montréal, QC, Canada), Arnaud Le Troter, Virginie Callot (Aix Marseille Université, CNRS, CRMBM/CEMEREM UMR 7339, Marseille, France), and Franck C. Sgard (IRSST, Montreal, QC, Canada)

Earplugs are a usual way to protect workers subjected to noise exposure. However, the efficiency of these hearing protection devices is often affected by induced discomforts. A factor suspected to impact both acoustical and physiological comfort attributes of earplugs is the deformation they apply on the ear canal walls. As the geometry of both open and occluded ear canal is difficult to obtain, the ear canal deformation due to earplug insertion is not trivial to evaluate. Current medical imaging techniques and image post-processing methods are promising tools to investigate this deformation. In a previous study of the authors, an approach using registration methods on medical images had been proposed to estimate the ear canal displacement field induced by earplug insertion. This approach had been validated in the case of computed tomography scans of a human-like artificial ear occluded by a controlled-shape custom molded earplug. In the present study, this approach is used to evaluate the ear canal displacement of a human subject for various earplug insertions (foam, pre-molded and custom made) in the case of magnetic resonance images. The computed displacement field shows noteworthy differences between each earplug and gives information on how and where the occluded ear canal deforms.

10:40

4aNS8. Sound localization performance of sample hearing protectors using standardized measurement methods. Eric R. Thompson (Battlespace Acoust., Air Force Res. Lab., 2610 Seventh St, B441, Wright-Patterson AFB, OH 45433, eric.thompson.28@us.af.mil) and Richard L. McKinley (Oak Ridge Inst. for Sci. and Education, Wright-Patterson AFB, OH)

Sound localization is an important aspect in the performance of many tasks. These tasks frequently require the use of hearing protection to mitigate the potentially adverse effects of noise exposure. ANSI/ASA working group S3/WG94 has described four standard methods for measuring the effect of head-worn devices on sound localization. Data using three different hearing protectors and open ear and using three of the four methods described will be presented. The analysis techniques used for each of the three methods will be described in detail including the removal of front-back reversals and assessment of fine resolution location discrimination. The performance of the three hearing protectors and the open ear condition will be compared and contrasted. Data will be presented on proportion of localization discrimination and front/back reversals using Method 1 as well as localization error performance using Methods 2, and visual target acquisition and response time using Method 3. The performance will also be compared across the three methods. The results show that the measurement methods described are viable for the assessment of the effects of HPDs on performance associated with localization.

Contributed Paper

11:00

4aNS9. Standard methods to measure the effects of head-worn devices on sound localization. Richard L. McKinley (Oak Ridge Inst. for Sci. and Education, 2610 Seventh St., AFRL/711HPW/RHCB, Wright-Patterson AFB, OH 45433-7901, rich3audio@aol.com) and Eric R. Thompson (Battlespace Acoust., Air Force Res. Lab., Wright-Patterson AFB, OH)

Four standard methods for measuring the effect of head-worn devices on sound localization in the horizontal plane have been developed by the working group S3/WG94. The objective was to establish methods that enable accurate, repeatable, and reliable measurement of sound localization performance for human listeners. The standard describes four measurement methods: (1) a low complexity, coarser method to estimate the proportion of location discrimination and front/back reversal errors using 8 loudspeakers; (2) a more complex, more robust method to measure localization error using 36 loudspeakers and a fine resolution response method; (3) a method to measure the functional impact of localization with degraded cues using 36 loudspeakers and an aurally guided visual search task; and (4) a method using 180 loudspeakers to precisely measure localization acuity. The methods will be described in detail.

Session 4aPA

Physical Acoustics, Noise, and ASA Committee on Standards: Sonic Boom I

Alexandra Loubeau, Cochair
Structural Acoustics Branch, NASA Langley Research Center, MS 463, Hampton, VA 23681

Joel B. Lonzaga, Cochair
Structural Acoustics Branch, NASA Langley Research Center, 2 N. Dryden St., B1208 MS 463, Hampton, VA 23681

Chair's Introduction—8:40

Invited Papers

8:45

4aPA1. Atmospheric turbulence effects on sonic boom signatures. Christopher Hobbs and Kevin Bradley (KBRwyle, 8580 Cinder Bed Rd., Ste. 1700, Lorton, VA 22079, chris.hobbs@wyle.com)

Two measurement campaigns were conducted as part of a study to quantify the effect of atmospheric turbulence on sonic boom propagation. Supersonic overflights were conducted at the NASA Armstrong Flight Research Center and the Kennedy Space Center to provide different atmospheric conditions for study. Measurements were made of the strength of the turbulence along with the height of the atmospheric boundary layer. Multiple microphone arrays recorded the sonic booms from instrumented aircraft. This work will present statistics detailing the variation of measured booms as a function of the strength of the turbulence and the propagation distance through the atmospheric boundary layer. [Work supported by NASA.]

9:05

4aPA2. Validation of numerical predictions of sonic boom metric variability due to turbulence. Trevor A. Stout and Victor Sparrow (Grad. Program in Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, stout.trevor.a@gmail.com)

Atmospheric turbulence scatters the sonic boom wavefront created by supersonic aircraft, causing random variations in the waveforms and spectra measured at the ground. To help assess the impact of turbulence, a time-domain algorithm based on the nonlinear KZK propagation equation has been developed which simulates turbulent fields via an approximate atmospheric theory. Two sets of recent supersonic overflight measurements at the NASA Neil A. Armstrong Flight Research Center and the Kennedy Space Center offer datasets which have been used to quantify the algorithm's utility to simultaneously model atmospheric turbulence, nonlinearity, and absorption. The field campaigns simultaneously recorded measurements of both sonic boom waveforms along linear microphone arrays and also turbulence quantities suitable for use in the algorithm. The algorithm's performance in predicting the measured variation of sonic boom metrics will be discussed. [Work supported by NASA via a subcontract from KBRwyle.]

9:25

4aPA3. Formulation of a Burgers' equation for shock waves in a turbulent atmosphere. 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)

The theoretical formulation of a Burgers' equation accounting for the attenuation and dispersion of shock waves due to scattering in a turbulent medium is presented. The physical effect treated here is in addition to the effects of geometric spreading, large-scale inhomogeneity, nonlinearity, viscous and thermal dissipation, and molecular vibrational relaxation. Although all Burgers' equations are inherently a one-way approximation, here the scattering effects are incorporated into the Burgers' equation through a correction to the small-amplitude sound speed. Such a correction is obtained from the full wave equation using a perturbative technique, and physically arises from multiple scattering of the signal due to small-scale inhomogeneities. The sound speed correction is complex, and is dependent on the frequency and the variance and length scale of the fluctuations. The real part of the correction gives rise to the dispersion of the signal while the imaginary part leads to signal attenuation. The latter is manifested by the increase in the rise time of the shock waves. The physical effect presented here is integrated into an existing sonic boom propagation code with minimal additional computational time. Numerical results will be compared with experimental data.

9:45

4aPA4. Experimentally derived turbulence filters applied to low amplitude sonic booms. Jacob Klos (Structural Acoust. Branch, NASA Langley Res. Ctr., 2 N. Dryden St., MS 463, Hampton, VA 23681, j.klos@nasa.gov)

Turbulence effects on low amplitude sonic booms are not currently well understood. In 2011, NASA funded an experiment to collect data using a large linear microphone array, consisting of 81 ground placed microphones spaced 125-feet apart, as a supersonic F-18B aircraft flew over the array. Most of the measurements were made during aircraft maneuvers designed to place a focus boom on the microphones. However, a few flights were straight and level passes of the aircraft over the array. Data from two of these level passes were used to estimate filter functions that describe the variation at each microphone relative to the mean response across the array. Two approaches were implemented, one is frequency domain and the second is time domain. The resulting filters can be applied to other non-turbulent boom waveforms to estimate the changes in either the spectrum or time domain signature due to the variations observed during these flights. Variation in loudness level both outdoors and indoors when these filters are applied to predicted signatures is presented. Both outdoors and indoors, the standard deviation of Perceived Level is only slightly smaller for low boom waveforms in comparison to N-waves.

10:05

4aPA5. Investigation of deturbing methods for sonic boom signatures. Janet L. Xu (Acoust., Penn State Univ., 1623 S. Ashwicken Ct., State College, PA 16801, jqx5082@psu.edu) and Victor Sparrow (Acoust., Penn State Univ., University Park, PA)

There is a desire to certify aircraft for supersonic flight using microphone measurements of the aircraft's sonic boom signature on the ground. If the signature propagates through the planetary boundary layer before it reaches the ground, the signature becomes distorted because of atmospheric turbulence, corrupting the measurement. Methods to remove the turbulence effect, or deturbing methods, currently exist but are restricted to typical N-wave shapes or knowledge of the turbulence structure beforehand. This work investigates methods of deturbing sonic boom signatures independent of the N-wave shape and not requiring prior knowledge of turbulence. Improvements are made to the averaging method and subtraction method of deturbing, and a new method is investigated which uses audio fingerprinting, an audio identification algorithm used by music identification apps such as Shazam. The effects of each deturbing method are described using a wide set of physical and subjective metrics. [Work supported by the FAA. The opinions, findings, conclusions, and recommendations expressed in this material are those of the authors and do not necessarily reflect the views of ASCENT FAA Center of Excellence sponsor organizations.]

10:25–10:40 Break

10:40

4aPA6. Sonic boom carpet widths for NASA's Low Boom Flight Demonstrator using realistic atmospheric profiles in the western hemisphere. William Doebler (The Penn State Univ., 201 Appl. Sci. Bldg., State College, PA 16802, wfd5057@psu.edu), Alexandra Loubeau (Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA), and Victor Sparrow (The Penn State Univ., University Park, PA)

NASA is developing and building a Low Boom Flight Demonstrator (LBFD) within the next five years to study public reaction to low amplitude, shaped sonic booms. Understanding the potential land area ensonified by sonic booms is critical for the upcoming community tests, and is not well known outside of using the 1976 US Standard Atmosphere which assumes zero wind. Using atmospheric profiles from NOAA's Climate Forecast System Reanalysis (CFSR) for a span of many months, the sonic boom carpet widths and Stevens Mark VII Perceived Levels (PL) at the edge of the carpet were calculated using NASA's sBOOM augmented Burgers Equation propagation code. The LBFD's near field pressure cylinder calculated using NASA's LAVA CFD code was used as the input to the sBOOM code. Atmospheric profiles were sampled from several regions with varying climate in the western hemisphere. Preliminary results indicate that sonic boom carpets will be widest during winter months, and will have highest carpet edge PL values during summer months. [Work supported by the NASA Education AS&ASTAR Fellowship Program.]

11:00

4aPA7. An improved Mach cut-off Model based on a 3-D ray tracing method and realistic atmospheric data. Zhendong Huang and Victor Sparrow (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, zfh5044@psu.edu)

This research aims to assess the possibilities for Mach cut-off flight over land. An improved Mach cut-off model has been developed using a 3-D ray tracing method. Based on the Integrated Global Radiosonde Archive (IGRA), a radiosonde dataset from the National Centers for Environmental Information (NCEI) consisting of radiosonde and pilot balloon observations, the influence of realistic atmospheric profiles and flight conditions on cut-off Mach numbers is examined. Examples are given for certain busy air routes in the United States. [Work supported by the FAA. The opinions, findings, conclusions and recommendations expressed in this material are those of the authors and do not necessarily reflect the views of ASCENT FAA Center of Excellence sponsor organizations.]

11:20

4aPA8. Mean flow atmospheric effects and their impact on sonic boom propagation. Sriram Rallabhandi (Aeronautics Systems Anal. Branch, NASA Langley, Rm. 190-25, Mailstop 442, NASA Langley Res. Ctr., Hampton, VA 23681, sriram.rallabhandi@nasa.gov)

One dimensional augmented Burgers equation has been generally used for nonlinear lossy sonic boom propagation. This equation models the effects of nonlinearities, loss mechanisms such as absorption and dispersion due to molecular relaxation and thermoviscous dissipation, and geometric spreading through ray tube areas including Blokhintzev scaling as well as atmospheric stratification.

However, in the presence of atmospheric winds, original terms in the augmented Burgers formulation do not account for the Doppler effects, where the observer is moving with the local flow rather than being fixed in space. Instead, wind is accounted for by updating the ray paths and effective speed of sound. Inclusion of mean flow wind effects in all terms of the augmented Burgers equation allows for an enhanced prediction capability that is closer to the underlying physics. This work will update sBOOM, an augmented Burgers' solver, to reflect the mean flow wind enhancements. The sonic boom ground signatures and other relevant data are compared against those obtained without using mean flow wind effects. The shock rise times, sonic boom duration and peak pressures are some variables that are expected to be different, resulting in differences in noise metrics. Such differences will be discussed and documented for cases which may include shaped low-boom as well as strong shock signatures.

11:40

4aPA9. Time-domain modeling of the wind effects on the nonlinearity and absorption of sonic booms. 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)

This paper discusses the derivation and solution, in the time domain, of a Burgers' equation for sonic boom propagation in a windy atmosphere. As the wind effects become important with increasing wind speed, the proper treatment of wind is necessary for sonic booms propagating through a jet stream as well as for those propagating to the stratosphere or thermosphere where stronger winds typically exist. The effects of wind are quantified in terms of the Doppler shift and of a parameter which measures the strength of the convection. The latter, referred to as the convective index, is a ratio of the local sound speed to the magnitude of the ray velocity and is unity in the absence of wind. Both Doppler shift and convective index are larger than unity if there is a component of the wind velocity opposite to the direction of the acoustic propagation. Consequently, the effective nonlinearity coefficient and effective absorption coefficients are larger for rays propagating opposite the wind direction than for rays propagating in the same direction. To demonstrate the significance of the derived Burgers' equation, the effect of a jet stream with a typical strength and profile on nominal sonic booms will be described.

THURSDAY MORNING, 10 MAY 2018 NICOLLET D2, 8:00 A.M. TO 11:35 A.M.

Session 4aPP

Psychological and Physiological Acoustics: Honoring Neal Viemeister's Contributions to Psychoacoustics

Andrew J. Oxenham, Cochair
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Chair's Introduction—8:00

Invited Papers

8:05

4aPP1. Temporal integration and multiple looks, revisited. Gregory H. Wakefield (Elec. Eng. and Comput. Sci. Dept., Univ. of Michigan, Ann Arbor, MI 48109, ghw@umich.edu)

Recent advances in Kullback-Leibler divergence and its generalizations have led to a renewed interest in the modeling of various detection and classification problems. Building from the concept of multiple looks, this paper considers adaptive detector and classifier structures that support stable, multi-stream perception. With respect to these structures, the phenomena of temporal integration and the distinctions between energetic and informational masking are discussed. [Work supported by ONR N00014-16-1-2560.]

8:25

4aPP2. Using psychoacoustics to examine a changing auditory system. Elizabeth A. Strickland (Speech, Lang., and Hearing Sci., Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, estrick@purdue.edu)

In the (long) time I was a student in Neal's lab, I learned a rigorous, programmatic approach to psychoacoustics that was grounded in signal detection theory. Much of the work at that time followed up on Neal's classic work in intensity discrimination and envelope processing, and how that might depend on processing within or across auditory channels. The research was tied to what was known

about physiological processing in the auditory system. Research from other laboratories at the time suggested that the cochlea might adjust in response to sound via efferent feedback, and a paper by Liberman used a physiological approach that could be easily translated to psychoacoustic experiments. This inspired a search for behavioral evidence of efferent feedback to the cochlea, and an array of experiments were performed looking for effects of contralateral sound on psychoacoustic measures. While those experiments did not provide compelling evidence of efferent feedback, they sparked an interest that has continued to be a focus of my research. This talk will cover the array of experiments done at that time, and consider the results in the context of more recent results from my laboratory and other laboratories. [Work supported by NIH(NIDCD)R01 DC008327.]

8:45

4aPP3. Effects of age and noise exposure on the representation of amplitude modulation. Christopher J. Plack (Dept. of Psych., Lancaster Univ., Ellen Wilkinson Bldg., Oxford Rd., Manchester M13 9PL, United Kingdom, chris.plack@manchester.ac.uk) and Samuele Carcagno (Dept. of Psych., Lancaster Univ., Lancaster, United Kingdom)

Age and lifetime noise exposure were used as predictors for electrophysiological and psychophysical measures of amplitude modulation processing. Frequency-following responses (FFRs) were recorded from 61 listeners (20 young, 23 middle-aged, 18 elderly) to 0.6- and 2-kHz, 75-dB SPL, carrier tones amplitude modulated at 100 Hz. A pink noise highpass-filtered at 3 kHz was included to mask basal cochlear contributions to the FFR. Sinusoidal amplitude modulation detection (AMD) thresholds were measured for the same listeners for a 2-kHz carrier (40 and 80 dB SPL) and for three modulation frequencies (25, 50, and 100 Hz). The carrier was presented in notched pink noise to limit off-frequency listening. Lifetime noise exposure was estimated using a structured interview. A regression model was used to determine the independent contributions of age and noise exposure, while controlling for audiometric threshold. FFR amplitude for the 0.6 kHz carrier showed a marked age-related decline. Age was also associated with a significant increase in AMD thresholds. Lifetime noise exposure was not significantly associated with either FFR amplitude or AMD threshold. The results suggest that age is a stronger predictor of modulation processing deficits than lifetime noise exposure.

9:05

4aPP4. Psychoacoustic considerations in hearing aid design and fitting. Brent Edwards (National Acoust. Labs., Australian Hearing Hub, Level 4, 16 University Ave., Macquarie University, NSW 2109, Australia, brent.edwards@nal.gov.au)

Hearing aid technology has advanced tremendously over the past two decades, with multiple technologies introduced to improve the hearing of wearers with hearing loss. While hearing science has informed their development, the design of these technologies and their fitting have been approached primarily from a speech-centric perspective. This talk will reflect on hearing aid technology from the perspective of the type of fundamental psychoacoustic research conducted by Neal Viemeister throughout his career. Additional psychoacoustic research that is needed to inform future hearing aid designs will also be identified. A cookie will be bet that Neal is surprised by the relationship between psychoacoustics and hearing aid design.

9:25

4aPP5. Context effects in modulation masking. Magdalena Wojtczak (Psych., Univ. of Minnesota, 1237 Imperial Ln, New Brighton, MN 55112, wojtc001@umn.edu)

Neal Viemeister is especially known for his contributions to our understanding of perception of amplitude modulation. Modulation masking and signal-to-noise ratio in the modulation domain have been shown to be strong predictors of speech intelligibility in noise. This study uses 40-ms amplitude-modulated tones embedded in a simultaneous 100-ms noise masker to show that for a given type of simultaneous masker, modulation masking can vary significantly depending on the type of stimulus that precedes the masked target. Three types of precursor were used to investigate these context effects, a 2-octave noise centered on the signal frequency (the same as the masker), a 7-component inharmonic complex tone, and a pure tone with a frequency equal to that of the target. All precursors had a duration of 400 ms and were presented in close temporal proximity at levels equal to those of the noise masker. The noise precursor produced on average a 7-8 dB [20log(m)] release from modulation masking. In contrast no change in the amount of masking and a small release of masking were observed for the pure-tone and complex-tone precursors, respectively. Mechanisms that may underlie the observed context effects will be discussed. [Work supported by NIH grant R01 DC015462.]

9:45–10:00 Break

Contributed Papers

10:00

4aPP6. Neal Viemeister's contributions to understanding the dynamic range of the auditory system. Brian C. Moore (Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB3 9LG, United Kingdom, bcjm@cam.ac.uk)

Most neurons in the auditory nerve show saturation of their firing rates in response to a tone with a moderate level presented at their characteristic frequency. However, humans can detect changes in intensity for levels up to 120 dB SPL. One explanation of this "dynamic range problem" is that intensity coding at high levels depends on the spread of cochlear excitation. To test this explanation, in 1974 Neal Viemeister measured the intensity discrimination of bursts of noise presented in a band-reject noise of fixed high intensity. He found that Weber's law held even at high levels. He concluded that spread of excitation is not necessary for the auditory system to maintain its large dynamic range. In 1983 Neal Viemeister showed that, over a wide range of intensities, subjects could detect small differences in the intensity of a high-frequency band of noise (6-14 kHz, a frequency range where phase locking is essentially absent) presented with a complementary band-reject noise, showing that phase locking is not essential for intensity discrimination at high levels. In 1988 Neal Viemeister showed theoretically that a localized rate-based intensity code using a small number of neurons can account for intensity discrimination over a wide range of levels.

10:15

4aPP7. Loudness of an auditory scene. William Yost (ASU, PO Box 870102, Tempe, AZ 85287, william.yost@asu.edu)

When multiple sound sources, each of equal level, are added together to create an auditory scene, the overall level of the scene grows with the increased number of sources despite the fact that the level of the individual sound sources remains the same. This experiment addresses a simple question: how does the overall perceived loudness of an auditory scene change as the number of sound sources in the scene changes. Sixteen Consonant-Vowels (CVs), spoken by ten males and ten females, were the sound stimuli. The CVs could be concatenated to produce nonsense "words" of different lengths. Different number ("n") of concatenated CVs could be presented at one time from a single loudspeaker located on the azimuth plane or from different loudspeakers. Listeners determined if "n + delta n" sound sources were louder than "n" sound sources. The individual levels of the concatenated CVs was another independent variable. Preliminary data suggest that when "n" is small (<4), the level of the individual sounds determines loudness judgements. When n is larger (n>4) the overall level of the simultaneously presented sounds determines loudness judgements. A full study will be presented. (Work supported by grants from NIDCD and Oculus VR, LLC.)

10:30

4aPP8. The contribution of Neal Veimeister to the modulation theory of hearing. Christian Lorenzi (Institut d'Etude de la Cognition, Ecole normale superieure, 29 rue d'Ulm, Paris, Ile de France 75005, France, christian.lorenzi@ens.fr)

Neal Viemeister has dedicated his scientific career to one of the most influential research programs in hearing sciences. This program aims to apply the modulation theory to auditory perception. This theory, inspired by telecommunication sciences, relies on the following assumptions: (i) communication sounds including speech and animal vocalizations convey salient and slow temporal modulation cues; (ii) the auditory system of species using communication sounds has evolved to optimize its responses to these modulation cues; (iii) the transmission of information conveyed by communication signals is constrained by the 'modulation transfer function' (MTF) of the transmission path (a room, a hearing aid,...). We will show how the pioneering work of Neal Viemeister on the 'temporal MTF' and his rigorous investigation of modulation perception by human listeners contributed to an in-depth understanding of the demodulation processes implemented in the peripheral and central auditory system. We will show how his work yielded to models of modulation processing that account for a large range of listening situations including speech intelligibility and texture perception. Finally, we will show how his work contributed to a better understanding of the perceptual consequences of ageing and cochlear damage, and a better control of information transmission via rehabilitation systems.

10:45

4aPP9. Amplitude modulation detection and modulation masking in school-age children and adults. Emily Buss, John Grose (The Univ. of North Carolina at 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)

Temporal resolution is often evaluated by measuring sensitivity to amplitude modulation (AM) as a function of modulation rate. Published data for AM detection on a noise carrier indicate that school-age children perform more poorly than adults, but thresholds increase with increasing AM rates above 50 Hz in a parallel fashion for both age groups. This result has been interpreted as reflecting adult-like temporal resolution in school-age children. However, this interpretation is complicated by the observation that inherent AM of the noise carrier can elevate AM detection thresholds in adults via modulation masking. Two studies with 5- to 11-year-olds and adults were carried out to better understand the development of AM detection with and without modulation masking. The first estimated thresholds for detecting 16-, 64- or 256-Hz sinusoidal AM on a 4.3-kHz carrier, a condition without modulation masking. Age effects were larger than previously observed with a noise carrier. The second study manipulated modulation masking by measuring detection of 16-Hz sinusoidal AM on a bandpass noise carrier (1.5—2.5 kHz), with and without additional masker AM at nominal rates of 6.3, 16, or 40.3 Hz. All listeners exhibited on-frequency modulation masking, with similar tuning to modulation rate for children and adults.

11:00

4aPP10. Predicting speech intelligibility based on the modulation spectrum and modulation frequency selectivity. Torsten Dau (Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, tdau@elektro.dtu.dk)

Speech intelligibility depends on factors related to the auditory processes involved in sound perception as well as on the acoustic properties of the sound entering the ear. A clear understanding of speech perception in complex acoustic conditions remains a challenge. Here, a computational modeling framework is presented that attempts to predict the speech intelligibility obtained by normal-hearing and hearing-impaired listeners in various adverse conditions. The model combines the concept of envelope frequency selectivity in the auditory processing of the sound with a decision metric that is based either on the signal-to-noise envelope power ratio or a correlation measure. The proposed model is able to account for the effects of stationary background noise, reverberation, nonlinear distortions and noise reduction processing on speech intelligibility. However, due to its simplified auditory preprocessing stages, the model fails to account for the consequences of individual hearing loss on intelligibility. To address this, physiologically inspired extensions of the auditory preprocessing in the model are combined with the modulation-frequency selective processing and the back-end processing that have been successful in the conditions tested with normal-hearing listeners. The goal is to disentangle the consequences of different types of hearing deficits on speech intelligibility in a given acoustic scenario.

11:15

4aPP11. Auditory enhancement and other context effects in normal, impaired, and electric hearing. Andrew J. Oxenham, Heather A. Kreft, and Lei Feng (Psych., Univ. of Minnesota, 75 E River Parkway, Minneapolis, MN 55455, oxenham@umn.edu)

Neal Viemeister's contributions to our understanding of auditory enhancement stood out because they combined novel and intriguing empirical data with a testable theoretical framework to explain the results. Thirty-five years later, the resultant "adaptation of inhibition" hypothesis remains the default explanation for auditory enhancement effects, and it can be used to account for both basic auditory context effects as well as speech context effects. Recent work in our lab has focused on comparing speech and non-speech context effects in people with normal and impaired hearing, as well as cochlear implants, in an attempt to elucidate the underlying neural mechanisms and to work towards compensating for any loss of context effects in clinical populations via signal processing. This presentation will provide a survey of recent progress in this area with examples from simultaneous masking, forward masking, and speech discrimination experiments. [Work supported by NIH grant R01DC012262.]

11:30–11:35 Panel Discussion

Session 4aSA

Structural Acoustics and Vibration, Physical Acoustics, and Engineering Acoustics: Acoustic Metamaterials I

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

9:00

4aSA1. Exploring heterogeneity and disorder In tunable elastic metamaterials. Paolo Celli, Behrooz Yousefzadeh, Chiara Daraio (California Inst. of Technol., Pasadena, CA), and Stefano Gonella (Univ. of Minnesota, 500 Pillsbury Dr. SE, Minneapolis, MN 55455-0116, sgonella@umn.edu)

The dominant paradigm in the design of elastic metamaterials revolves around periodic arrays of identical resonating elements, which are known to allow subwavelength locally-resonant bandgaps. In this work, we deliberately explore the wave manipulation capabilities of metamaterial configurations that are characterized by significant heterogeneity and disorder. Heterogeneity is here intended as the coexistence of multiple types of resonators tuned to resonate at distinct frequencies. Disorder refers to the arrangement of the resonators, which is randomized in space. The metamaterials of choice are thin elastic sheets endowed with heterogeneous populations of tunable telescopic pillars whose resonant characteristics can be agilely tuned through simple manual operations. The configurability of our experimental platform allows to seamlessly fabricate and compare a multitude of specimens, thus allowing to extract some empirical yet general rules. We first document how heterogeneity can lead to broadband mechanical filtering. More specifically, we illustrate how randomized spatial arrangements consistently outperform their functionally graded counterparts in terms of maximum achievable bandgap width. Our investigation shows that the design space of mechanical metamaterials can be stretched beyond the limits imposed by order and homogeneity, thus highlighting the engineering significance of the emerging concepts of organized disorder and design modularity.

9:20

4aSA2. Design, optimization, and fabrication of mechanical metamaterials for vibration control. Timothy F. Walsh, Chris Hammetter (Computational Solid Mech. and Structural Dynam., Sandia National Labs., PO Box 5800, MS 0380, Albuquerque, NM 87185, tfwalsh@sandia.gov), Michael B. Sinclair, Harlan Shaklee-Brown (Electron., Optical, Nano, Sandia National Labs., Albuquerque, NM), Joe Bishop (Solid Mech., Sandia National Labs., Albuquerque, NM), and Wilkins Aquino (Civil Eng., Duke Univ., Durham, NC)

Harsh shock and vibration environments are common in engineering applications. Mechanical metamaterials are showing significant potential as candidates for controlling wave propagation and isolating sensitive structural components, but require proper design of their complex microstructures. In this talk we will present time and frequency-domain strategies for Partial Differential Constrained (PDE)-constrained design optimization of locally resonant elastic/acoustic metamaterials. We will present a variety of notch filter resonators and split ring cylinder/sphere geometries that can be easily optimized for wave control applications, along with fabrication details involving multi-material additive manufacturing. A frequency-domain approach will be presented for band-gap or notch filter materials, and a time-domain strategy for transient shock environments. As the metamaterial structures typically involve concentrated masses and elastic connections, an optimization strategy involving 3D simple shapes will be presented. [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. Department of Energy's National Nuclear Security Administration. With main facilities in Albuquerque, N.M., and Livermore, C.A., Sandia has major R&D responsibilities in national security, energy and environmental technologies, and economic competitiveness.]

9:40

4aSA3. Extraordinary wave phenomena in active acoustic metamaterials. Romain Fleury (EPFL, EPFL - STI - LWE, ELB 033 - Station 11, Lausanne 1015, Switzerland, romain.fleury@epfl.ch)

Non-Hermitian sonic metamaterials are artifical acoustic media that exploit gain and loss as an extra degree of freedom to induce novel physical effects that cannot be obtained with lossless or trivial lossy systems. In this talk, we will review our recent theoretical and experimental results about one-dimensional non-Hermitian acoustics, highlighting the ability of active metamaterials to overcome limitations of passive metamaterials by engineering the acoustic power flow at the microscopic scale. We will discuss various experiments employing a set of electroacoustic resonators in a pipe, engineered to provide tailored distributions of acoustic gain or loss, and discuss the potentials of such systems in terms of sound manipulation. Altogether, our work points out the large potential and rich physics of non-Hermitian acoustic metamaterials.

Contributed Papers

10:00

4aSA4. Impact wave attenuation using Maxwell-type oscillators in dissipative elastic metamaterials. Sagr Alamri, Bing Li, and Kwek Tze Tan (Mech. Eng., Univ. of Akron, 302 E Buchtel Ave, Akron, OH 44325, sma114@zips.uakron.edu)

This work studies the development of a dissipative elastic metamaterial with single and dual-Maxwell type resonator for stress wave mitigation. Mass-spring-damper elements are used to model and analyze the mechanism of wave dissipation effect on the vibration characteristics. It is found that broadband wave attenuation region can be obtained and expanded to a wider range by properly utilizing interactions from resonant motions and visco-elastic effects of the Maxwell-type oscillators. In addition, numerical verifications are conducted for various cases, and excellent agreement between theoretical and numerical frequency response functions are obtained. The design of this dissipative metamaterial system is further applied for blast wave mitigation. By means of the bandgap merging effect that is induced by the Maxwell-type oscillator, the passing blast wave can be almost completely suppressed in the low-frequency range. A significantly improved performance of the proposed dissipative metamaterials for stress wave mitigation is verified in both time and frequency domains.

10:15–10:30 Break

10:30

4aSA5. Broadband asymmetric wave transmission in dissipative acoustic/elastic metamaterials with diatomic oscillators. Bing Li, Sagr Alamri, and Kwek Tze Tan (Dept. of Mech. Eng., The Univ. of Akron, The University of Akron, Akron, OH 44325, bingli@uakron.edu)

Asymmetric acoustic/elastic wave transmission have attracted increasingly attention recently. However, the propagation unidirectionality is always confined to a very narrow frequency band. This paper presents the development of a dissipative acoustic/elastic metamaterial with diatomic oscillators to realize "one-way" transmission in a broadband range. The effect of various types of dissipative oscillators on the asymmetric wave transmission is theoretically investigated and systematically discussed. By virtue of the surficial vibrational modes induced by the alternately arranged diatomic resonators and the merging effect of dissipative dashpot, the asymmetric transmission band is significantly broadened. Numerical results in frequency and time domains are verified using both lattice structures and continuum models. Preliminary experimental verification is further conducted. The broadband asymmetric transmission can be theoretically predicted and mathematically manipulated by careful design of the unit size parameters and deliberate selection of material properties, which could be potentially beneficial for applications in directional transducer and noise control.

10:45

4aSA6. Underwater acoustic ground cloak development and demonstration. Peter Kerrian, Amanda Hanford, Benjamin Beck, and Dean Capone (Appl. Res. Lab., Pennsylvania State Univ., PO Box 30, State College, PA 16804, ald227@psu.edu)

Acoustic ground cloaks, which conceal an object on a rigid surface, utilize a linear coordinate transformation to map the flat surface to a triangular void by compressing space into two triangular cloaking regions consisting of a homogeneous anisotropic acoustic metamaterial. Transformation acoustics allows for the realization of a coordinate transformation through a reinterpretation of the scale factors as a new material in the original coordinate system. An underwater acoustic ground cloak was constructed from perforated steel plates and experimentally tested to conceal an object on a pressure release surface. The perforated plate acoustic ground cloak successfully cloaked the scattered object. There was excellent agreement between the phase of the surface reflection and cloak reflection with a small amplitude difference. Above 15 [kHz], the cloaking performance decreased as the effective material parameters of the perforated plate metamaterial deviated from the required material parameters.

11:00

4aSA7. Comparative study of impurities in lithium tantalite and lithium niobate. Chandrima Chatterjee (Phys. and Astronomy, The Univ. of MS, University, Lewis Hall, Rm. 108, University, MS 38677), Daniel Miller (Phys. and Astronomy, The Univ. of MS, Oxford, MS), and Igor Ostrovskii (Phys. and Astronomy, The Univ. of MS, MS, iostrov@phy.olemiss.edu)

S 38677 The point defects in ferroelectrics may influence the physical properties of phononic crystals $LiTaO_3$ (LT) and $LiNbO_3$ (LN). The initial wafers, periodically poled (PP) structures of LT with 1-mm-domains, and PP-LN with 0.45-mm-domains are investigated. Photoluminescence (PL) is taken at room temperature 310-nm -excitation. The slit width in optical spectrometer is set to 1 nm. The PL spectra of LT reveal multiple impurities including Fe^+,W, Th, Cs, Kr, Cu, Xe. Unlike PL from LN, there are no lines of the F-center, Ar and some other elements. The intensities of PL-lines in both LT and LN demonstrate a saw-type periodicity along the direction normal to the Z-axis. For instance, the PL line of Fe^+ defect in LT varies $\pm$ 15% with periodicity about 200 $\pm$ 40 nm. In the case of PP LT phononic crystals, the difference in Fe^+ line amplitude remains about the same; however, the periodicity becomes equal to a ferroelectric domain length of 1 mm. There is a difference between the two periodicities of charged defects in PP-LT and PP-LN; the positions of maxima in PL-lines are close to the interdomain walls in PP-LN, but it is not the case for PP-LT. Possible practical applications are discussed.

11:15

4aSA8. On the behavior of acoustic/elastic metamaterials with anisotropic mass density. mehran jaberzadeh, Bing Li, and Kwek Tze Tan (Mech. Eng., The Univ. of Akron, Akron, OH 44325, mehran.jaberzadeh@gmail.com)

The effect of anisotropic mass density in acoustic/elastic metamaterials on wave propagation is presented in this research. The use of microstructures to achieve anisotropic physical properties in metamaterials is an intensive field of study. In this work, a two-dimensional (2D) 'mass-in-mass'-spring lattice system is utilized to achieve anisotropic effective mass density in two orthogonal principal directions.In each direction, the effective mass density is frequency-dependent and can be "effectively negative" if it falls within the frequency bandgap region. A 2D numerical continuum model,-based on a recently developed cantilever-in-mass model, is further presented and examined to study how wave input angles affect 2D wave propagation. Results show that wave attenuation is dependent on both input frequency and wave input angle in a 2D metamaterial. This study demonstrates a case whereby wave attenuation is achieved,by selecting wave input frequency in the bandgap region to enact "negative effective mass density". This study also illustrates another case when wave attenuation is obtained for positive anisotropic mass density. This behavior is achieved by directing wave propagation to transverse direction at a specific input wave angle of 60°. Both numerical calculations of mass-spring lattice model and continuum cantilever-in-mass model show good agreement.

11:30

4aSA9. Non-reciprocal wave phenomena through pump-signal wave interaction in discrete systems with asymmetric subwavelength geometry. Samuel P. Wallen, Benjamin M. Goldsberry, and Michael R. Haberman (Appl. Res. Labs. - The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, walles@uw.edu)

Acoustic non-reciprocity has been shown to enable a plethora of effects analogous to phenomena seen in quantum physics and electromagnetics, such as immunity from back-scattering, unidirectional band gaps, and topologically protected states, which could lead to the design of direction-dependent acoustic devices. One class of material that holds promise as a means to achieve acoustic non-reciprocity is the "Willis medium," which exhibits strain-momentum coupling owing to asymmetry and non-local effects in the material microstructure. This work considers a method for obtaining non-reciprocal elastic wave propagation in a system with sub-wavelength asymmetry inspired by asymmetric Willis microstructures. Non-reciprocity is achieved through application of a slowly-varying pump wave that acts as a spatio-temporal modulation of the material, which is modeled as a discrete system with geometric nonlinearity. The modulation generates momentum bias for a fast-varying signal wave with pump-signal interaction enabled by weak nonlinearity associated with the variation of subwavelength asymmetric geometry. It is shown that momentum bias may be generated with a pump wave acting transverse to the direction of propagation, which may facilitate experimental realizations. [This work was supported by the National Science Foundation.]

THURSDAY MORNING, 10 MAY 2018 NICOLLET A, 8:00 A.M. TO 12:00 NOON

Session 4aSC

Speech Communication: Perception and Processing of Voice and Speech (Poster Session)

Terrin N. Tamati, Chair

Department of Otorhinolaryngology / Head and Neck Surgery, University Medical Center Groningen, Hanzeplein 1, Groningen 9700RB, Netherlands

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 authors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

4aSC1. The impact of perceptual load on natural and synthetic speech perception. Adriana Ojeda, Ethan Kutlu, and Ratree Wayland (Dept. of Linguist, Univ. of Florida, P.O. Box 115454, Gainesville, FL 32611-5454, ojedaae13@ufl.edu)

Previous work has shown that the availability and the weighting of different types of information (e.g., lexical, segmental/subsegmental, and metrical prosody) for speech segmentation are gradiently affected by perceptual and cognitive load. Specifically, it has been shown that severe energetic masking increases reliance on the acoustic, sub-segmental, context-dependent coarticulation information and decreases reliance on lexical information (e.g., Mattys *et al.*, 2009). In this study, the effects of energetic masking on the intelligibility of synthetic and natural speech (controlled for lexical and metrical prosody) will be compared. Since fine acoustic detail linked to coarticulation is lacking in synthetic speech in comparison to naturally produced speech, energetic masking should result in greater reliance on lexical information than coarticulatory information on synthetic speech processing while the opposite will be true for natural speech. *Reference* Mattys, S. L., Brooks, J., & Cooke, M. (2009). Recognizing speech under a processing load: Dissociating energetic from informational factors. Cognitive Psychology, 59, 203-243.

4aSC2. Asymmetries in vowel perception arise from phonetic encoding strategies. Matthew Masapollo (Boston Univ., 677 Beacon St., Boston, MA 02215, mmasapol@bu.edu), Lauren Franklin, James Morgan (Cognit., Linguistic & Psychol. Sci., Brown Univ., Providence, Rhode Island), and Linda Polka (McGill Univ., Montreal, QC, Canada)

Directional asymmetries in vowel discrimination studies reveal that speech perceivers (both adult and infant) are biased toward extreme vocalic articulations, which lead to acoustic vowel signals with well-defined spectral prominences due to formant convergence. These directional effects occur with vowel stimuli presented in either the acoustic or the visual modality and are independent of specific linguistic experience. Current research is focused on elucidating the perceptual processes underlying this universal vowel bias. In the present investigation, the inter-stimulus interval (ISI) in AX discrimination tasks for unimodal acoustic and visual vowels was manipulated (500 ms vs. 1000 ms) in order to examine whether asymmetries are present under experimental conditions that reduce demands on attention and working memory. Subjects discriminated either video-only or audio-only tokens of naturally-spoken English [u] and French [u] which differ in their degree of visible lip-rounding and proximity between F1 and F2. We observed robust asymmetries with these stimuli with English- and French-speaking adults in earlier studies using a relatively long ISI (1500 ms). The present results demonstrated that asymmetries in both auditory and visual vowel discrimination are diminished in the short ISI conditions, suggesting that this vowel bias derives from phonetic encoding processes, rather than general psychophysical processes.

4aSC3. Formant pattern and spectral shape ambiguity in vowel synthesis: The role of fundamental frequency and formant amplitude. Thayabaran Kathiresan (Dept. of Computational Linguist, Univ. of Zurich, Andreasstrasse 15, Zurich, Zurich 8050, Switzerland, thayabaran.kathiresan@uzh.ch), Dieter Maurer (Inst. for the Performing Arts and Film, Zurich Univ. of the Arts, Zurich, Switzerland), Heidy Suter (Inst. for the Performing Arts and Film, Zurich Univ. of the Arts, Zürich, Switzerland), and Volker Dellwo (Dept. of Computational Linguist, Univ. of Zurich, Zurich, Switzerland)

When investigating formant pattern and spectral shape ambiguity in Klatt synthesis, an earlier study showed that the perceived vowel quality of Standard German vowel sounds can be changed by varying fundamental frequency only [Maurer *et al.* (2017). J. Acoust. Soc. Am. **141**(5):3469-3470]. In this follow-up study, the previous original synthesis experiment was repeated twice, firstly, with fundamental frequencies (f_o) of the corresponding sounds lowered by one octave, and secondly, with different ratios of the first and second formant amplitudes. Here, the role of the f_o range and the formant amplitudes for the investigation of formant pattern and spectral shape ambiguity in vowel synthesis was further examined. The same five phonetic expert listeners that participated in the previous experiment also identified all of the newly synthesised sounds in a multiple-choice identification task. Results revealed that the perceived vowel quality only changes for f_os above 200 Hz and that, for back vowels, the ratio of the formant amplitudes used in the Klatt synthesis also affects vowel recognition. Thus, the results of the experiments confirm earlier indications of a non-systematic relation between f_o or pitch and formant patterns or spectral envelopes for vowel recognition.

4aSC4. Sinewave vowel sounds: The role of vowel qualities, frequencies and harmonicity of sinusoids, and perceived pitch for vowel recognition. Dieter Maurer (Inst. for the Performing Arts and Film, Zurich Univ. of the Arts, Toni-Areal, Pfingstweidstrasse 96, Zurich 8031, Switzerland, dieter.maurer@zhdk.ch), Heidy Suter (Inst. for the Performing Arts and Film, Zurich Univ. of the Arts, Zürich, Switzerland), Thayabaran Kathiresan, and Volker Dellwo (Dept. of Computational Linguist, Univ. of Zurich, Zurich, Zurich, Switzerland)

In the literature, the recognition of sinewave vowels replicating statistical formant patterns is reported as impaired when compared to natural sounds. However, the corresponding formant simulating sinusoids were harmonically unrelated, with synthesised signals only accidentally being quasi-periodic, and vowel confusion was indicated to relate to vowel height. Involving five phonetic expert listeners, the present study tested vowel and pitch recognition of three sinusoid replicas based on statistical F_1-F_2-F_3 patterns of the Standard German closed and mid-open vowels /i-y-e-ø-o-u/ for women, "corrected" approximations of these patterns with harmonically related sinusoids, and harmonical patterns with fixed first and third sinusoids, yet varying only the second sinusoid so as to effect a change in harmonic relation. The results showed strong vowel confusions for mid-open but only limited confusions for closed vowels. Additional effects on vowel recognition were indicated to concern harmonicity and specific frequencies of the sinusoids, and perceived pitch (range recognised = 165-440Hz). Thus, sinewave replications of formant frequencies seem to represent perceived vowel qualities not per se but only in relation to specific vowel qualities, sinusoid configuration and pitch, supporting earlier claims of spectral representation of vowel quality as being non-systematic and pitch-related.

4aSC5. Trial-to-trial variability in talkers' fundamental frequencies restrains spectral context effects in vowel categorization. Ashley Assgari and Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, 317 Life Sci. Bldg., Louisville, KY 40292, ashley.assgari@louisville.edu)

Perception of a given sound is influenced by spectral properties of surrounding sounds. For example, listeners perceive /ɪ/ (low F1) more often when following sentences filtered to emphasize high-F1 frequencies, and perceive /ɛ/ (high F1) more often following sentences filtered to emphasize low-F1 frequencies. These biases in vowel categorization are known as *spectral contrast effects* (*SCEs*). When preceding sentences were spoken by acoustically similar talkers (low variability in mean f0), SCEs biased vowel categorization, but sentences spoken by acoustically different talkers (high variability in mean f0) biased vowel categorization significantly less (Assgari *et al.*, 2016 ASA). However, it was unclear whether these effects varied due to local (trial-to-trial) or global (across entire block) variability in mean f0. Here, the same sentences were arranged to increase/decrease monotonically in mean f0 across trials (low local variability) or vary substantially from trial-to-trial (high local variability) with equal global variability. On each trial, listeners heard a sentence filtered to add a low-F1 or high-F1 spectral peak to bias categorization of a subsequent vowel (/ɪ/-/ɛ/ continuum). Sentences with low local variability in mean f0 biased vowel categorization significantly more than sentences with high local variability. Relevance to studies of talker normalization will be discussed.

4aSC6. Listeners compensate for prosodic position when categorizing word-final obstruents. Jeremy Steffman (Linguist, UCLA, 3125 Campbell Hall, Los Angeles, CA 90095-1543, jsteffman@g.ucla.edu)

This study explores how perceptual sensitivities to contextual variability extend to prosodically induced variation. In English, vowel duration is reliably longer preceding a voiced obstruent, as opposed to a voiceless obstruent, and listeners use preceding length as a cue to obstruent voicing (e.g. Raphael, 1972). Segmental duration also co-varies systematically with prosodic position, being longer phrase-finally (e.g. Turk & Shattuck-Hufnagel, 2007). In this study, we tested the extent to which listeners' categorization of word-final obstruents is altered by the prosodic position of the target. Participants heard a continuum that varied only in vowel length, and categorized stimuli as either *"coat"* or *"code"*. Prosodic position in a carrier phrase was manipulated by splicing the target word into either a phrase-final or phrase-medial context. Results show listener expectations about phrase final lengthening mediate categorization, with significantly longer vowel durations required in phrase-final position for a *"code"* response. The results are discussed in terms of other top-down influences on segmental categorization (e.g. lexical information), and implications for further work on the prosody-phonetics interface.

4aSC7. Evaluating mechanisms underlying nonspeech context effects in coarticulatory compensation. Navin Viswanathan (Speech-Language-Hearing, Univ. of Kansas, 1000 Sunnyside Ave, Lawrence, KS 66045, navin@ku.edu)

Human speech listeners overcome variability in the speech signal due to different speakers, rates, phonetic contexts etc., by demonstrating context-appropriate perceptual shifts. From a general auditory perspective, perceptual systems heighten contrastive spectral and temporal properties of the acoustic signal to help listeners cope with variability (Diehl *et al.*, 2004). In this study, I focus on a spectral contrast account of perceptual coarticulatory compensation and evaluate the claim that listeners cope with coarticulation by tracking spectral averages across multiple segments (Holt, 2005) while ignoring spectral variability (Holt, 2006). In Experiment 1, using tone analogue contexts (Lotto & Kluender, 1998) I created multi-tone contexts such that the global spectral average of each trial was pit against the frequency of the final tone that immediately preceded the target speech. Interestingly, nonspeech context effects were determined by the frequency of the final tone rather than the global trial average. A follow up experiment 2 indicated that contexts with identical overall spectral averages can produce different effects on following speech depending on the final tone. These findings taken together call into question the assumed mechanisms underlying spectral contrast.

4aSC8. The role of segment probability in perception of speech sounds. Seongjin Park, Maureen Hoffmann, Priscilla Z. Shin, and Natasha L. Warner (Dept. of Linguist, Univ. of Arizona, P.O. Box 210025, Tucson, AZ 85721, seongjinpark@email.arizona.edu)

This study investigates how listeners utilize segment probability in speech perception. In Warner *et al.* (2014), participants identified gated fragments of all combinations of two segments in English (e.g., diphones /hə/, /dt/, /ab/, /iɛ/). That data was further analyzed to study the influence of segment probability. Overall probability of the second segment (e.g., the probability of /t/ in English), and conditional probability of the second segment given the first segment (e.g., the probability of /b/ given preceding /a/) were calculated from the CMU pronunciation dictionary. There were significant correlations between accuracy of segment perception and both overall and conditional probability of second segments at early gates of the diphone, when few or no acoustic cues to the second segment were available (for example, perception of /b/ in /ab/ by one-third of the way through the /a/). However, when participants had enough acoustic information, the correlation was smaller or non-significant. The present study shows that English listeners utilize probability in identifying English segments when few acoustic cues are available, whereas Dutch listeners in a matched previous study on perception of Dutch (Warner *et al.* 2005) utilized probability only for a few unusual cases.

4aSC9. A replication of competition and prosodic effects on spoken word recognition. Natasha L. Warner, Genesis G. Hernandez, Seongjin Park (Dept. of Speech and Hering Sci., University of Arizona, Box 210071, Tucson, AZ 85721, genesishernandez@email.arizona.edu), and James M. McQueen (Radboud Univ., Nijmegen, Netherlands)

Despite the absence of clear and reliable word boundary markers, listeners recognize words in spoken sentences. A previous study tested a spoken-word recognition model, Shortlist, by asking speakers of British English to identify real words within nonwords (McQueen, Norris & Cutler, 1994). Some words were embedded within the onsets of longer words with either a weak-strong (WS) or strong-weak (SW) stress pattern (e.g., "mess" in /dəmɛs/, the onset of "domestic," "sack" in /sækɹɪf/, the onset of "sacrifice), and other words were embedded without a real word onset with either a WS or SW pattern (e.g., /nəmɛs/ or /mɛstəm/ for "mess" and /sækɹək/ or /kləsæk/ for "sack"). The original study reported both competition effects (e.g. competition from "domestic" hindered recognition of "mess) and prosodic effects (e.g. the stress in "mess" facilitated segmenting it from the preceding context). This current study aimed to replicate these results in American-English. Despite the different listener population and dialect, pilot results for the current study confirm both types of effect. This data will be used to test a new American English version of the Shortlist-B model of spoken word recognition (Norris & McQueen, 2008).

4aSC10. Cortical entrainment to speech under competing-talker conditions: Effects of cognitive load due to presentation rate and task difficulty. Paul Iverson, Jieun Song, and Holly Bradley (Univ. College London, 2 Wakefield St, London WC1N 1PF, United Kingdom, p.iverson@ucl.ac.uk)

Previous work has demonstrated that, when listeners attend to a target talker and ignore a distractor, their neural activity is more strongly entrained to the amplitude envelope of the target speech. Moreover, target-talker entrainment under these conditions can be higher for second-language listeners than native listeners, which might be due to the greater focused attention required to understand second-language speech. The present study manipulated the cognitive difficulty of the task in an attempt to make native-language listeners have entrainment results more like those of second-language listeners. Listeners heard simultaneous sentences spoken by male and female speakers, and had to detect semantically anomalous catch trials. The sentences had high or low semantic predictability (low-predictability sentences require more lexical processing to evaluate meaning) and were presented with short or long intervals between sentences (0.5 or 3.0 s, with shorter intervals forcing listeners to make a quicker evaluation of meaning). The preliminary results suggest that native-language listeners indeed have higher target-talker entrainment under more cognitively difficult conditions, suggesting that cognitive load may modulate focused attention to the target speech signal.

4aSC11. Electrophysiological measures of listening effort and comprehension: Speech in noise, vocoders, and competing talkers. Anna Exenberger and Paul Iverson (Univ. College London, 2 Wakefield St, London WC1N 1PF, United Kingdom, p.iverson@ucl.ac.uk)

This study explored the relationship between signal degradation and electrophysiological measures of speech processing, in order to understand how these measures relate to intelligibility and listening effort. Subjects heard two talkers presented to different ears, with the target talker presented in quiet, three levels of speech-shaped noise (+3 dB, -0.5dB, -4dB SNR), or a 14-channel noise vocoder. The speech materials were simultaneous sentences that varied in final-word predictability; subjects were asked to monitor the sentences for catch trials (i.e., semantically anomalous final words). The results demonstrated that cortical entrainment to the amplitude envelope of the target was higher than for the distractor, as found in previous studies. Entrainment and lexical processing are robust with increasing noise, then fall when the stimulus becomes very poorly intelligible. Unexpectedly, results from the vocoded condition indicate no entrainment advantage for the target speaker, even when this condition is intelligible.

4aSC12. The time course of word recognition in younger and older adults: Signal degradation and lexical challenge. Kristin Van Engen (Psychol. and Brain Sci., Washington Univ. in St. Louis, One Brookings Dr., Campus Box 1125, Saint Louis, MO 63130-4899, kvanengen@wustl.edu), Avanti Dey (Psychol. and Brain Sci., Washington Univ. in St. Louis, New York, NY), Nichole Runge, Mitchell Sommers (Psychol. and Brain Sci., Washington Univ. in St. Louis, Saint Louis, MO), Brent Spehar (Otolaryngol., Washington Univ. in St. Louis, St. Louis, MO), and Jonathan E. Peelle (Otolaryngol., Washington Univ. in St. Louis, Saint Louis, MO)

As listeners age, speech recognition often becomes more challenging, especially in noisy environments. Older adults also appear to have more difficulty resolving lexical competition than do younger adults. Relatively little is known, however, about how such signal- and lexicon-related challenges interact to affect the timing of word recognition and how such effects might change across the lifespan. In this study, we use a visual world paradigm to investigate the effects noise, word frequency, and phonological neighborhood density on the time course of spoken word recognition in young and older adults. On each trial, listeners saw four items on a computer screen and heard "Click on the" followed by a target word that corresponded to one of the images. None of the other items on the display were phonologically or semantically related to the target. Eye-tracking results show that the time course of word recognition is predicted by age, noise, frequency, density, and several of their interactions. In general, high-frequency words and words from dense phonological neighborhoods were fixated more quickly, while noise and age slowed recognition. These results support a framework in which lexical and acoustic factors co-determine the cognitive challenges associated with speech perception.

4aSC13. Semantic context modulates intelligibility advantage of clear speech in temporally compressed sentences. Valeriy Shafiro (Commun. Disord. & Sci., Rush Univ. Medical Ctr., 600 S. Paulina Str., AAC 1012, Chicago, IL 60612, valeriy_shafiro@rush.edu), Rajka Smiljanic, and Sandie Keerstock (Linguist, Univ. of Texas at Austin, Austin, TX)

Sentences produced in a clear speech speaking style are typically more intelligible than sentences produced in a conversational style. However, factors that lead to the intelligibility difference are not fully understood. This study investigated the role of semantic context in intelligibility of temporally compressed clear and conversational speech. Grammatically correct English sentences, which were either semantically meaningful or semantically anomalous, were presented to 70 normal-hearing listeners in clear or conversational speaking style. The sentences were temporally compressed to produce seven different syllabic rates (6-12), matched across clear and conversational sentences to control for the original durational differences between clear and conversational speech. Results for semantically anomalous sentences indicate no differences in the rate of intelligibility decline across compression rates for clear and conversational speech. In contrast, for semantically meaningful sentences, the rate of intelligibility decline was smaller for the semantically meaningful clear-speech sentences over semantically anomalous conversational-speech sentences. At the highest compression rate of 12 syllables per second, intelligibility of semantically meaningful clear-speech sentences (~68% correct) was similar to that of semantically anomalous sentences at 9 syllables per second. These findings suggest that semantic context can modulate the effect of speaking style on the intelligibility of temporally compressed sentences.

4aSC14. Ganong shifts for noise- and sine-vocoded speech continua in normal-hearing listeners. Brian Roberts and Robert J. Summers (Psych., School of Life and Health Sci., Aston Univ., Birmingham B4 7ET, United Kingdom, b.roberts@aston.ac.uk)

Speech perception involves interpreting incoming sensory information in the context of stored linguistic knowledge. Lexical bias is the tendency to perceive an ambiguous speech sound as a phoneme that completes a word rather than a non-word; the more ambiguous the auditory signal, the greater the reliance on lexical knowledge. For example, a speech sound ambiguous between [g] and [k] is more likely to be perceived as [g] when preceding

[Ift] and as [k] when preceding [Is]. The magnitude of this between-context difference—the Ganong shift—also increases when the listener experiences high cognitive load, limiting the resources available to process acoustic-phonetic information. In this study, an eight-step series between monotonized natural tokens of [gI] and [kI] was created using morphing software. Two continua—"gift"-"kift" and "giss"-"kiss"—were derived by adding [ft] or [s] to each step. High-resolution noise- (16-band) and sine-vocoded (32-band) versions were derived from these continua. Ganong shifts were substantially larger for sine- than for noise-vocoded stimuli, despite the greater number of channels for the former and similar slopes for the identification functions, perhaps reflecting the higher cognitive load associated with the relative spectral sparsity and less natural timbre of the sine-vocoded stimuli. [Work supported by ESRC.]

4aSC15. Across-ear integration of acoustic-phonetic information under competitive conditions: Effects of attenuation increase in the presence of an extraneous formant. Robert J. Summers and Brian Roberts (Psych., School of Life and Health Sci., Aston Univ., Birmingham B4 7ET, United Kingdom, r.j.summers@aston.ac.uk)

Previous research using consonant-vowel syllables where one ear receives the first formant (F1) and the other receives the second and third has shown that dichotic release from masking allows F2 + F3 to remain effective speech cues even after substantial attenuation. This study used three-formant analogues of natural sentences and extended the approach to include competitive conditions. Target formants were presented dichotically (F1 + F3; F2), either alone or accompanied by an extraneous competitor for F2 (i.e., F1 ± F2C + F3; F2) that listeners must reject to optimize recognition. In experiment 1, F2C was absent and F2 was attenuated in 6-dB steps (range: -6-48 dB). Intelligibility was unaffected until attenuation >30 dB; F2 still provided useful information at 48-dB attenuation. In experiment 2, F2 was attenuated in 12-dB steps (range: 0-36 dB). F2C was created by inverting the F2 frequency contour and using the F2 amplitude contour without attenuation. When F2C was present, 12-dB attenuation was sufficient to cause some loss of intelligibility; the effect was large when attenuation ≥24 dB. This interaction suggests that some mandatory across-ear spectral integration occurs, such that informational masking arising from F2C rapidly swamps the acoustic-phonetic information carried by F2 as its relative level is attenuated. [Work supported by ESRC.]

4aSC16. Multimodal perception of interrupted speech and text by younger and older adults. Rachel E. Miller and Daniel Fogerty (Commun. Sci. and Disord., Univ. of South Carolina, 1229 Marion St, Columbia, SC 29201, remiller@email.sc.edu)

The current study aimed to identify how partial auditory (speech) and visual (text) information are resolved for sentence recognition by younger and older adults. Interrupted speech was created by periodically replacing speech with silent intervals at various interruption rates. Likewise, interrupted text was created by periodically replacing printed text with white space. Speech was further processed by low-pass filtering the stimulus spectrum at 2000 Hz. Speech and text stimuli were interrupted at 0.5, 2, 8, and 32 Hz and presented in unimodal and multimodal conditions. Overall, older listeners demonstrated similar performance to younger listeners on the unimodal conditions. However, there was a marked decrease in performance by older listeners during multimodal testing, suggesting increased difficulty in integrating information across modalities. Performance for both groups improved with higher interruption rates for both text and speech in unimodal and multimodal conditions. Listeners were also able to gain benefit from most multimodal presentations, even when the rate of interruption was mismatched between modalities. Supplementing speech with incomplete visual cues can improve sentence intelligibility and compensate for degraded speech in adverse listening conditions. However, the magnitude of multimodal benefit appears mitigated by age-related changes in integrating information across modalities. [Work supported, in part, by NIH/NIDCD.]

4aSC17. Musical training and the perception of phonetic detail in a shadowing task. Grace Ji Yan Tsang (Communicative Sci. and Disord., New York Univ., 665 Broadway, Fl. 9, New York, NY 10012, gjt238@nyu.edu), Edmund L. Dana, Morwaread M. Farbood (Music Technol., New York Univ., New York, NY), and Susannah V. Levi (Communicative Sci. and Disord., New York Univ., New York, NY)

Phonetic convergence, or shadowing, is the phenomenon in which people unintentionally and temporarily change phonetic details of their speech to sound more similar to another talker. Previous research has examined effects of the shadower and of the target speaker. In the current study, we examine listener contributions on the perception of phonetic convergence. Two hundred and sixty listeners completed an AXB perception task in which they were asked to determine whether the first or third stimulus was a better imitation of the middle stimulus. Listeners provided information about previous instrumental and vocal training. Preliminary examination of the results reveals that listeners with more musical training (in the form of instrumental or vocal experience) lead to better accuracy. Results from the study suggest that there is a link between musical training and detection of fine-grained phonetic details in speech.

4aSC18. Is the target of imitation directly acoustic or a pattern within a speaker's phonetic system? Rebecca Scarborough (Linguist, Univ. of Colorado, 295 UCB, Boulder, CO 80309, rebecca.scarborough@colorado.edu) and Kuniko Nielsen (Linguist, Oakland Univ., Rochester, MI)

Speakers may converge phonetically with another talker to whose speech they are exposed, resulting in reduced acoustic distance between interlocutors. However, it is unclear whether the target of imitation is raw acoustics or a linguistic pattern. Zellou *et al.* (2016) present a case that distinguishes these possibilities: after listening to a model talker whose speech was manipulated to reduce coarticulatory vowel nasality (measured acoustically in A1-P0), participants' nasality also decreased in post-test productions, relative to their baseline. However, the model speaker had naturally low A1-P0 (correlated with high nasality), and even after the manipulation to raise A1-P0, it was lower than all participants'. If the target of imitation is acoustic, participants *diverged* from the model (raised A1-P0). But if the target of imitation is a linguistic pattern (decreased coarticulatory nasality), participants *converged*. The current study uses an AXB perceptual task to determine whether it is acoustically more similar items (more nasal; speakers' baseline) or linguistically more similar items (less nasal; post-test) that listeners judge as "more similar" to the model talker's production. A generalized linear mixed model showed a significant preference for post-test items (55.4%, $z = 5.19$), indicating that linguistic similarity is the basis for listeners' similarity judgments.

4aSC19. Power and phonetic accommodation. Auburn Lutzross, Andrew Cheng, Alice Shen, Eric Wilbanks, and Azin Mirzaagha (Dept. of Linguist, Univ. of California Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94704, azshen@berkeley.edu)

Theories of speech accommodation claim that phonetic convergence to one's interlocutor is automatic (Goldinger, 1998; Shockley *et al.*, 2004). However, there is evidence that individual traits and social factors modulate accommodation so that speakers may even diverge from their interlocutor (Babel, 2010; Bourhis & Giles, 1977). This study investigates how talker's personal sense of power and power relative to the interlocutor ("interpersonal power") affect phonetic accommodation. Personal power was manipulated through a thought experiment in which participants described a time they felt powerful or powerless. Interpersonal power was manipulated through randomly assigned roles in a recorded interview with a confederate where the Inventor (powerless) pitched an entrepreneurial idea to the Investor (powerful). Participants were considered to have converged if the difference in pitch between the participant and confederate had decreased in post-interview versus pre-interview recordings. We found that though interpersonal power does not influence the direction or degree of accommodation, personal power does. Speakers with lower personal power diverged from their interlocutor ($t = -2.1389$, $p = 0.04$), indicating that an individual's self-perception has a stronger effect on accommodation than the established roles in the interaction.

4aSC20. The effects of lexical and phonological factors on talker processing. Sandy Abu El Adas and Susannah V. Levi (Communicative Sci. and Disord., New York Univ., 665 Broadway, 9th Fl., New York, NY 10012, sandyabu@nyu.edu)

Previous research has shown that listeners are better at processing talker information in their native language compared to an unfamiliar language, a phenomenon known as the Language Familiarity Effect. Several studies have explored the cause of this effect. Some have argued that it is tied to the lexicon and the ability to comprehend, while others have suggested that it is the familiarity with the phonology. In the current study, we use an AX discrimination task to test these two factors simultaneously by manipulating lexical status (words/nonwords) and phonotactic probability (high/low). We also test individual differences in reading ability, as poor phonological awareness skills are linked to reading impairment. Twenty-four native speakers of American English completed the AX task and a battery of phonological awareness tasks. Reaction time results revealed an interaction between lexical status and phonotactic probability: words with high phonotactic probability were processed faster than words with low phonotactic probability, but no difference was found for the nonwords. Sensitivity (A') revealed no effects of lexical status or phonotactic probability, but did reveal that listeners with higher reading scores were more sensitive to talker differences.

4aSC21. Effects of training on the perception of talker information in degraded speech. Terrin N. Tamati (Dept. of Otorhinolaryngology / Head and Neck Surgery, Univ. Medical Ctr. Groningen, Hanzeplein 1, Groningen 9700RB, Netherlands, t.n.tamati@umcg.nl)

A fundamental property of human speech perception is its robustness in a wide range of listening conditions. Listeners are able to successfully recognize and understand speech under a wide range of adverse and challenging conditions, such as in noise, reverberation, or multi-talker babble, and with noise vocoding. Noise vocoding, a manipulation that reduces spectral resolution, has been shown to result in less accurate speech comprehension and also poorer perception of indexical information (e.g., talkers' voices). However, speech comprehension quickly improves with exposure, as listeners learn to better extract information from the systematically degraded speech. The extent to which the learning of noise-vocoded speech transfers to other talkers or tasks is still largely unknown. The current study investigated whether training, consisting of transcribing sentences or identifying talkers' voices from sentences, leads to improvements in the perception of talker information. Preliminary results suggest that both training tasks resulted in improved perception of talker information, as participants learned general information about the degradation. Larger improvements were observed after talker identification training, which focused attention on talker differences. Differences between training tasks, and implications for cochlear implant users, will be discussed. [Funding: VENI Grant (275.89.035) from the Netherlands Organization for Scientific Research (NWO).]

4aSC22. Perceptual similarity judgments of voices: Effects of talker and listener language, vocal source acoustics, and time-reversal. Kristina Furbeck, Emily J. Thurston, Jessica Tin, and Tyler K. Perrachione (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, tkp@bu.edu)

Listeners demonstrate reliable perceptual biases favoring voices speaking their native language versus those speaking a foreign language. This "language-familiarity effect" for voice processing has been found even for perceptual similarity judgments: Voices speaking listeners' native language sound less alike, even when those recordings have been rendered incomprehensible via time-reversal. Here, we sought to replicate and extend this finding of linguistic effects on voice similarity. Native English- and Mandarin-speaking listeners (both $N=40$) rated the perceptual similarity of voices speaking English or Mandarin (both $N=20$) for either time-reversed or forward speech. Both listener groups tended to find Mandarin-speaking voices more dissimilar, but this effect was reduced for English-speaking listeners, especially for forward speech. Perceptual similarity judgments of voices were always highly correlated between listener groups and between forward/time-reversed speech. Acoustic measurements (voice fundamental frequency mean and variance, local jitter, harmonics-to-noise ratio, speech rate, and formant dispersion) were also made to ascertain how listeners' perceptual similarity judgments were related to speech acoustics and whether these relationships differed under time-reversal or across listeners' or talkers' native language. Overall, these data suggest that, while native language may influence perceptual dissimilarity of voices, the magnitude of these effects tend to be very small.

4aSC23. Impact of talker adaptation on speech processing and working memory. Sung-Joo Lim, Jessica Tin (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., 610 Commonwealth Ave, Auditory Neurosci. Lab., Boston, MA 02215-2422, sungj.m.lim@gmail.com), Barbara Shinn-Cunningham (Biomedical Eng., Boston Univ., Boston, MA), and Tyler K. Perrachione (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., Boston, MA)

Talker adaptation is known to facilitate immediate speech recognition. However, it is unclear whether and how talker adaptation also facilitates working memory for speech. Using electroencephalography (EEG) during a delayed recall of digit span task, we investigated whether talker adaptation facilitates working memory performance. We also investigated whether neural alpha oscillatory power reflects facilitatory effects of talker adaptation. Listeners encoded sequences of seven randomly ordered digits, recalling them after a 5-s retention. Digit sequences were spoken by either a single talker or multiple talkers, and were presented at either faster or slower speech rates (0-ms vs. 500-ms inter-digit intervals). Overall, listeners responded faster and more accurately for single-talker sequences compared to multiple-talker sequences. Especially for the faster presentation rate, listeners were more efficient (faster and more accurate) in recalling sequences spoken by a single talker. For the faster presentation rate, processing digit sequences spoken by single vs. multiple talkers also elicited reduced alpha power during both speech encoding and working memory retention. These results suggest that talker adaptation reduces cognitive effort during both speech encoding and memory retention, thereby producing more efficient working memory for speech information, especially when listeners process speech rapidly.

4aSC24. Re-examining the effect of top down processing on voice perception. Ashley Quinto, Kylee Kim, and Susannah V. Levi (Communicative Sci. & Disord., New York Univ., 665 Broadway, 9th Fl., New York, NY 10012, anq203@nyu.edu)

The current study aimed to replicate a recent study conducted by Narayan, Mak, & Bialystok (2016) that found effects of top-down linguistic information on a talker discrimination task by comparing four conditions: compounds (day-dream), rhymes (day-bay), reverse compounds (dream-day), and unrelated words (day-bee). The original study used both within- and across-gender pairs and same and different trials were analyzed separately, obscuring possible response biases. Narayan *et al.* found graded performance between the four conditions, but some results were likely to have been influenced by the use of across-gender trials in the different-talker condition. In the current study, only female speakers were used and results were analyzed with signal detection theory (sensitivity and bias measures). Results revealed that participants were faster to respond to rhyming pairs than the three other conditions. In addition, participants were significantly more sensitive to talker differences in rhyming pairs than unrelated pairs, but no other conditions differed. Participants were more biased to respond "same" in the rhyme and compound conditions than in the unrelated condition. These results demonstrate a partial replication of the Narayan, Mak, & Bialystok (2016) findings, suggesting an interaction between linguistic and talker information during speech perception.

4aSC25. Perception of femininity in normally phonated and whispered speech. Nichole Houle and Susannah V. Levi (Communicative Sci. and Disord., New York Univ., 665 Broadway, New York, NY 10012, nh1473@nyu.edu)

Many transgender women seek out voice and communication therapy to support in their transition from their biological gender to presenting as their gender identity. This has led to an increased need to examine the perception of gender and femininity to develop evidence-based therapy practices. In this study, we explored perception of femininity in normally phonated and whispered speech. Transgender male-to-female, cismale, and cisfemale speakers

were recorded producing hVd words. Naïve listeners rated femininity using a visual-analog scale. The results revealed that listeners rated speakers more ambiguously in whispered speech than normally phonated speech. Within-group analyses were conducted to further examine how speaker characteristics (height, age, mean f0, duration) contributed to perceptions of femininity. While there was a significant effect of mean f0 within the normally phonated condition for all groups, none of the other speaker or acoustic cues consistently predicted listener ratings of femininity across speaker groups.

4aSC26. Investigating the influence of listener attitudes and expectations on the intelligibility of Hindi-influenced English. 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)

Phoneme recognition and speech intelligibility can be affected by purely social factors, such as beliefs about or attitudes toward speakers of varying races and ethnicities (e.g., Babel & Russell, J. Acoust. Soc. Am. **137** [2015]). The current study examined whether implicit and explicit attitudes about people from South Asian (SA) affect the intelligibility of Hindi-influenced English (HIE). Three measures were used: an Implicit Association Test (IAT) to measure general associative attitudes toward SA individuals; an intelligibility task in which listeners repeated HIE and North American English (NAE) sentences in noise, accompanied by pictures of SA or white individuals; and an ethnographic interview to acquire qualitative data about stereotypes, communicative practices with nonnative speakers, and other language attitudes. Data collection is ongoing in a more ethnically diverse location (Vancouver, BC) and a more homogenous one (Minneapolis, MN). Drawing from past studies on socioindexical influences on nonnative speech, we predict that individuals who harbor negative attitudes and stereotypes about SA people will perceive HIE as less intelligible than those without such beliefs. The results of this study will inform the growing body of literature regarding the ways in which socioindexical factors influence speech recognition across individuals with diverse ethnicities and language varieties.

4aSC27. The perception of sexual orientation through speech: Generational change. Lily Obeda and Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu)

Previous research has shown that listeners can perceive the sexual orientation of some lesbian, gay, and bisexual (LGB) people at a greater than chance levels based on the acoustic-perceptual characteristics of their speech (Munson *et al.*, J. Phonetics [2006], Pierrehumbert *et al.*, J. Acoust. Soc. Am. [2004]). The cues to these judgments were the pronunciation of specific sounds (i.e., /æ/, /ɛ/, /oʊ/, /u/, /s/), rather than global speech characteristics (like f0 range or formant-frequency scaling). In the years since those studies were conducted, societal attitudes toward LGB people have changed substantially. Moreover, ongoing sound changes have affected the pronunciation of many of the sounds that cued judgments of sexual orientation from speech. The current study examines whether these changes affect the perception of sexual orientation through speech. The perception experiments described in Munson *et al.* are being redone using two groups of listeners. One group is matched in age to subjects whose data were reported previously (i.e, currently 18-30 years old). The second group that is matched in birth year to the subjects from the earlier study (i.e., currently 33-45 years old). Ongoing analyses compare these new data are compared to those from Munson *et al.*

4aSC28. Gender typicality in children's speech, reconsidered: Effects of stimulus composition on listener judgments. McKalya Beaulieu, Emily Larson, and Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu)

Listeners rate the speech of boys and girls as young as four years old as sounding gendered: boys are rated as sounding boy-like and girls as girl-like (Perry *et al.*, J. Acoust. Soc. Am. [2001]). Recent research found that the extent to which boys' speech sounds boy-like is correlated with measures of their gender identity and expression (Li *et al.*, J. Phonetics [2016], Munson *et al.*, J. Acoust. Soc. Am. [2015]). Munson *et al.* found that boys with a diagnosis of gender identity disorder [GID] were rated as sounding less boy-like than boys without GID. Munson *et al.*'s experiment used only a small number of girls' productions as filler items. The current study examined listeners' ratings of the gender typicality of speech of boys with GID and both boys and girls without GID. Significant differences in sex-typicality ratings were found between the two groups of boys. Boys with GID elicited ratings intermediate to those for boys and girls without GID. However, the differences between boys with and without GID were much smaller than those in Munson *et al.*, suggesting that the sex distribution in the stimulus set can affect ratings of the sex typicality of children's voices.

4aSC29. Children's ability to benefit from fundamental frequency and vocal tract length differences during speech-in-speech recognition. Mary M. Flaherty, Lori Leibold (Ctr. for Hearing Res., Boys Town National Res. Hospital, 555 N. 30th St., Omaha, NE 68106, maryflah@buffalo.edu), and Emily Buss (UNC Chapel Hill, Chapel Hill, NC)

The present study evaluated whether children's speech-in-speech recognition benefits from differences in fundamental frequency (F0) or vocal tract length (VTL) between the target and masker talkers' voices. Children, like adults, can benefit from a sex mismatch between competing talkers, but the relative contribution of individual voice characteristics to children's improved speech-in-speech understanding is unknown. In this study, we first tested children's ability to use differences in either F0 or VTL between target and masker speech to evaluate the independent influence of these cues on speech-in-speech recognition. Then F0 and VTL differences were combined to determine whether cue redundancy would reduce the child/adult differences observed in these contexts. Sentence recognition thresholds were measured in a two-talker speech masker. All stimuli were recorded by the same female talker. F0 and VTL of the target sentences were manipulated using the pitch-synchronous overlap-add method. Preliminary results suggest a prolonged developmental trajectory in the ability to use F0 or VTL in isolation, but indicate that the combination of these cues benefits children at an earlier age compared to when either cue is presented in isolation. Adults showed the greatest benefit from the F0-only manipulation, showing no additional benefit when F0 and VTL were combined.

4aSC30. Effects of speech competitors on encoding features of novel word-object pairs. Katherine M. Simeon and Tina M. Grieco-Calub (Commun. Sci. & Disord., Northwestern Univ., 2240 Campus Dr., Frances Searle Bldg. Rm 2-381, Evanston, IL 60208, ksimeon@u.northwestern.edu)

Fast-mapping is the ability to map novel word-object pairs with very few exposures (Carey & Bartlett, 1978), and is thought to underlie language acquisition. However, language learning often happens in natural environments that contain competing sounds. Previous studies examining whether competing sounds affect fast-mapping yielded inconsistent results, likely due to methodological differences. The present study adds to this body of work by testing the effect of competing sounds on the depth of encoding novel word-object pairs during a fast-mapping task. Three-to-four-year-old children performed a fast-mapping task in quiet and in the presence of a two-talker speech competitor presented at a +2 dB signal-to-noise ratio. In each condition, children were trained on three novel word-object pairs, whereby the object was verbally labeled and associated with related attributes while moving across a computer screen (e.g., "This *modi* can *bounce*"). Children were then tested on their recognition of each word-object pair. Children associated words to objects equally well in quiet and speech competitor conditions, but recalled fewer object attributes when speech competitors were present. Results suggest that speech competitors disrupt encoding of the semantic features of word-object pairs. This presentation will highlight how different methodologies demonstrate varying effects of competing sounds on fast-mapping.

Session 4aUW

Underwater Acoustics, Signal Processing in Acoustics, and Physical Acoustics: High Performance Computing Applications to Underwater Acoustics

Ying-Tsong Lin, Cochair
Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543

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

Chair's Introduction—8:30

Invited Papers

8:35

4aUW1. High performance computing methods for nonlinear Bayesian uncertainty quantification. Jan Dettmer (Dept. of Geoscience, Univ. of Calgary, 2500 University Dr. NW, Calgary, AB T2N 1N4, Canada, jan.dettmer@ucalgary.ca), Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and Charles W. Holland (Appl. Res. Lab., Penn State Univ., State College, PA)

Bayesian uncertainty quantification (UQ) for nonlinear inverse problems requires the application of numerical integration (sampling) to estimate the posterior probability density since no closed-form expressions exist. Nonlinear problems are common in geophysics, in particular, in applications of studying the seabed with sound. The computational cost of applying UQ can be daunting and we present several approaches that improve efficiency for large inverse problems. Fundamentally, Bayesian sampling includes both fine-grained parallelism in the forward model (data prediction) as well as coarse grained parallelism in the sampling. Fine grained parallelism can be addressed efficiently by implementation on massively parallel accelerators, such as graphics processing units, and application to reflection coefficient computation is shown. The coarse grained parallelism of sampling is ideally addressed by traditional parallelization with message passing across a cluster of computers. We consider implementation of parallel sampling algorithms that scale efficiently to 10^3 computer cores. Finally, computational efficiency is closely tied to parametrization efficiency, which is addressed by self-adapting parametrizations of unknown complexity that lead to parsimonious representations of complex environments with few parameters. These three aspects of efficiency will be illustrated for several inverse and imaging problems in seabed acoustics and seismology. [Funded by the Natural Sciences and Engineering Research Council of Canada.]

8:55

4aUW2. Massively parallel structural acoustics in Sierra-SD. Gregory Bunting, Scott T. Miller, and Timothy F. Walsh (Computational Solid Mech. and Structural Dynam., Sandia National Labs., 14211 Vista CT NE, Albuquerque, NM 87108, gbuntin@sandia.gov)

Sierra-SD is a massively parallel finite element application for structural dynamics and acoustics. Problems on the order of 2 billion degrees of freedom and running on as up to one hundred thousand distributed memory cores have been solved. Sierra-SD offers a wide range of modern acoustic capabilities. Unbounded problems can be solved with either an absorbing boundary condition, infinite elements, or perfectly matched layers. Problems can be solved in the time domain, frequency domain, or as a linear or quadratic eigenvalue problem. Structural acoustics problems can be solved with monolithic strong coupling, or a Multiple Program Multiple Data (MPMD) coupling with the linear and nonlinear structural Sierra applications. One way handoff is available from other applications in the Sandia National Laboratories' Sierra Mechanics suite, enabling loading through the Lighthill tensor from incompressible fluid codes to calculate far field noise. Real world applications include ship shock loading and vibration of reentry bodies. [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. Department of Energy's National Nuclear Security Administration under contract DE-NA-0003525.]

9:15

4aUW3. Some examples of the application of high-performance computing for time-domain full-wave simulations in underwater acoustics using a spectral-element method. Paul Cristini, Dimitri Komatitsch, and Alexis Bottero (CNRS-LMA, 4, Impasse Nikola Tesla, CS40006, Marseille 13013, France, cristini@lma.cnrs-mrs.fr)

Being able to perform accurate time-domain full-wave simulations in underwater acoustics is important within the context of complex environments. Among the numerical methods that can be used to perform such types of simulations, the spectral-element method is a method of choice because it can take advantage of the possibilities offered by high-performance computing. This capability is very important because accurate numerical simulations require a high computational power. In this presentation we will show some examples of the use a spectral-element method for underwater acoustics applications. 2D simulations as well as 3D simulations will be considered for various configurations, illustrating the interest of performing such simulations. Shallow water propagation, deep water propagation as well as diffraction by objects in complex environments will be analyzed.

9:35

4aUW4. Finite element modeling for ocean acoustics applications using high performance computing. Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

Finite element models provide solutions to the Helmholtz equation for ocean acoustics applications, which are exact on the order of the discretization. However, to achieve convergence the discretization must be on the order of several elements per acoustic wavelength. For large-scale ocean acoustics applications, and small (high-frequency) acoustic wavelengths, the number of elements required often exceeds millions. In addition, in each element, the basis set decomposition required for finite element modeling increases the number of degrees of freedom, and for problems involving elastic structures, additional degrees of freedom are required to describe the elastodynamic acoustics equation. Therefore, for a large-scale ocean acoustics problem, the number of degrees of freedom can reach into the tens of millions. This requires large memory caches on computing systems. The number of degrees of freedom can be reduced by using wavenumber decomposition techniques at the expense of running many models. In this talk, large-scale ocean acoustics finite element modeling will be reviewed from an historical perspective, the current state of modeling will be discussed from the in light of available computing resources and the future of finite element modeling will be considered. [Work supported by ONR, Ocean Acoustics.]

Contributed Papers

9:55

4aUW5. Numerical modeling with high performance computing of seismic waves for complex marine environments: Benchmarking with laboratory experiments. Bence Solymosi, Nathalie Favretto-cristini, Vadim Monteiller, Paul Cristini (CNRS-LMA, 4, Impasse Nikola Tesla, CS40006, Marseille 13013, France, cristini@lma.cnrs-mrs.fr), Borge Arntsen (NTNU, Trondheim, Norway), Dimitri Komatitsch (CNRS-LMA, Marseille, France), and Bjorn Ursin (NTNU, Trondheim, Norway)

Numerical simulations are widely used for forward and inverse problems in seismic exploration to investigate different wave propagation phenomena. However, the numerical results are hard to be compared to real seismic measurements as the subsurface is never exactly known. Using laboratory measurements for small-scale physical models can provide a valuable link between purely numerical and real seismic datasets. We present a case study for comparing ultrasonic data for a complex model with spectral-element and finite-difference synthetic results. The small-scale model was immersed in a water tank. Reflection data was recorded with piezoelectric transducers using a conventional pulse-echo technique. We paid special attention to the implementation of the real source signal—and radiation pattern—in the numerical domain. It involved a laboratory calibration measurement, followed by an inversion process. The model geometry was implemented using a non-structured mesh for the spectral-element simulations. The comparisons show a very good fit between synthetic and laboratory traces in general, and the small discrepancies can be assigned mostly to the noise present in the laboratory data.

10:10–10:25 Break

10:25

4aUW6. Integrated ocean acoustics and dynamics modeling with computer clusters. Ying-Tsong Lin, Timothy F. Duda, Weifeng G. Zhang, and Arthur Newhall (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu)

Underwater sound propagation in the ocean can be influenced by a variety of physical oceanographic processes, including ocean currents, eddies, internal gravity waves, tides, fronts, instabilities, etc. Our group at the Woods Hole Oceanographic Institution has been developing several numerical simulation frameworks to integrate ocean acoustic and dynamical models and implement them on computer clusters. Most of the current dynamical ocean models run on clusters, and we will review our integration approaches in this talk and also point out the requirement for nesting model domains and interpolations. Different parallelization schemes will be introduced to speed up a variety of acoustic numerical models, for example, ray-tracing, normal modes, couple modes and Parabolic-Equation approximation models. Examples of 3D sound propagation in a number of ocean environments (canyons, slopes, continental shelves and basins) with different dominating ocean processes will be demonstrated. Model performance improvement and limitation will be discussed. [Work sponsored by the ONR.]

10:40

4aUW7. Broadband numerical simulations of scattering from rough surfaces. Derek R. Olson (Oceanogr., Naval Postgrad. School, 201 Appl. Sci. Bldg, University Park, PA 16802, olson.derek.r@gmail.com)

Numerical simulation of rough surface scattering via discretization of the Helmholtz-Kirchhoff integral equation (HKIE) is a powerful, yet computationally expensive method to study rough surface scattering. In this work, simulations are performed using Fourier synthesis of the numerical solution of the HKIE using the boundary element method (BEM). Each frequency component is solved using BEM employing a good approximation of a plane wave as an incident field. The surface length requirements on this incident field depend on angle, and all simulations are performed using plane waves whose angular width is less than one degree, and whose mean grazing angle is greater than 20 degrees. This requirement resulted in problems with over 22,000 degrees of freedom and approximately 3000 individual frequencies. This broadband technique is employed to study the dependence of the scattered intensity on surface parameters as well as the bandwidth of the transmitted pulse. Simulations are performed both on surfaces with truncated power law spectra, as well as rippled surfaces with power-law roughness superimposed.

10:55

4aUW8. Scattered field formulation applied to 3D models for buried targets. Aaron M. Gunderson, Anthony L. Bonomo, and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 99164-2814, aaron.gunderson01@gmail.com)

Calculation of the acoustic scattering from buried targets in a seafloor environment is a task ideally suited for finite element software. A complication arises though when the target's environment is modeled as two infinite half spaces of water and sediment, each surrounded by perfectly matched layers (PML's) to satisfy the Sommerfeld radiation condition. Each half space requires a surrounding PML tailored to its material properties, resulting in abrupt discontinuities in PML structure at the corners where the sand-water interface meets the edges of the physical domain. Acoustic energy incident on these corners results in nonphysical reflections directed back into the physical domain. It has previously been shown in a 2D axisymmetric geometry [Zampolli *et al*., JASA **122**, 2007] that using a scattered field formulation, where the incident, reflected, and transmitted fields are applied as background fields, bypasses the discontinuity and eliminates these spurious reflections. Here, the theory is extended to full 3D geometries, allowing the scattering from 3D buried objects to be evaluated and tested against experimental and analytical benchmark results. Once complete, the 3D model template is easily modified to consider any target at any level of burial. [Work supported by Applied Research Laboratories IR&D and ONR, Ocean Acoustics.]

11:10

4aUW9. Sound radiation from a cylinder in shallow water by FEM and PE method. Zhiwen Qian, Dejiang Shang (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin Eng. University UnderWater Acoust. Fl., Harbin, Hei Longjiang 150001, China, 15846595689@163.com), and Yuanan He (Collage of underwater Acoust. Eng., Harbin Eng. Univ., Beijing, China)

There is no reliable research method for the acoustic radiation prediction of elastic structures in shallow water at present. Based on the finite element method coupled with the parabolic equation, the theoretical model for structure acoustic radiation in shallow water in low frequency is established in this paper. The vibro-acoustic characteristics of elastic cylinder influenced by with upper and lower fluid boundaries is calculated and analyzed. The study shows that when the cylindrical shell approaches the sea surface or bottom, the coupled frequency is higher or lower respectively than that of the shell immersed in the free field. When the diving depth reaches a certain distance range, the coupled frequency tends to be the same as that in free field. The acoustic field radiated from an elastic shell in Pekeris waveguide is similar to that from a point source in low frequency, but significant differences exist between them in high frequency. The structural source can't be equivalent to a point source. The research results also show that the method proposed in this paper has good adaptability to the type of structure geometry and shallow water waveguide, and the calculation efficient and accurate are also very high.

11:25–11:40 Panel Discussion

THURSDAY AFTERNOON, 10 MAY 2018 GREENWAY F/G, 1:30 P.M. TO 3:15 P.M.

Session 4pBAa

Biomedical Acoustics: Therapeutic Ultrasound and Bioeffects

Emad S. Ebbini, Chair
Electrical and Computer Engineering, University of Minnesota, 200 Union St. SE, Minneapolis, MN 55455

Contributed Papers

1:30

4pBAa1. *In vivo* ultrasound thermal ablation controlled using echo decorrelation imaging. Mohamed A. Abbass, Neeraja Mahalingam, Krishna S. Krothapalli (Biomedical Eng., Univ. of Cincinnati, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, abbassma@mail.uc.edu), Syed A. Ahmad (Surgery, Univ. of Cincinnati, Cincinnati, OH), and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH)

The utility of echo decorrelation imaging feedback for real-time control of *in vivo* ultrasound thermal ablation was assessed in rabbit liver with VX2 tumor. High-intensity focused ultrasound (HIFU) and unfocused (bulk) ablation was performed using 5 MHz linear image-ablate arrays. Treatments comprised up to nine lower-power sonications, followed by up to nine higher-power sonications, ceasing when the average cumulative echo decorrelation within a control region of interest exceeded a predefined threshold (-2.3, $\log_{10}$-scaled echo decorrelation per millisecond, corresponding to 90% specificity for tumor ablation prediction in previous *in vivo* experiments [Fosnight *et al*., Ultrasound Med. Biol. 43:176-186, 2017]). This threshold was exceeded in all cases for both HIFU ($N = 13$) and bulk ($N = 8$) ablation. Controlled HIFU trials achieved significantly smaller average lesion width compared to previous uncontrolled trials. Both controlled HIFU and bulk ablation trials required significantly less treatment time than uncontrolled trials. Prediction of local liver and VX2 tumor ablation using echo decorrelation was tested using receiver operator characteristic curve analysis, showing prediction capability comparing favorably with uncontrolled trials. These results indicate that control using echo decorrelation imaging may reduce treatment duration and increase treatment reliability for *in vivo* thermal ablation.

1:45

4pBAa2. Patient-specific large-volume hyperthermia in the liver for ultrasound-enhanced drug delivery from thermosensitive carriers. Brian Chu, Robin Cleveland, and Constantin Coussios (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, St. Cross College, St. Giles, Oxford, Oxfordshire OX1 3LZ, United Kingdom, brianchu100@googlemail.com)

Ultrasound-mediated mild hyperthermia in the range of 39C to 42C is a promising technique for noninvasive targeted release of thermally sensitive carriers in cancer therapy. The need to sustain a temperature rise of a few degrees evenly throughout a large volume currently limits this therapy to regions with a large accessible acoustic window. Furthermore, the need to

4p THU. PM

continuously scan clinically available highly focused therapeutic ultrasound transducers typically used for ablation limits the volume that can be maintained at this temperature for tens of minutes, and greatly extends treatment time per unit volume. In addition, bone structures, such as the ribs, significantly restrict accessible target locations in areas such as the liver and play a significant role in preventing this therapy from entering into clinical practice. In order to overcome these limitations, we present a combined 3D full wave finite element modelling and experimental approach to design a sectored lens that can be placed on focused transducers to enable direct mild heating of a larger volume in the liver while accounting for patient-specific aberration in the prefocal path. [Work supported by the RCUK Digital Economy Programme, grant number EP/G036861/1 (Oxford Centre for Doctoral Training in Healthcare Innovation).]

2:00

4pBAa3. The effects of high intensity focused ultrasound on biofilms formed by *Pseudomonas aeruginosa*. Lakshmi D. Bharatula (School of Chemical and Biomedical Eng., Nanyang Technolog. Univ., Singapore, Singapore), Enrico Marsili, Scott Rice (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)

Bacterial infections are increasingly difficult to treat due to their growing resistance to antibiotics. Most of these bacterial infections form a biofilm that limits the effectiveness of the antibiotic. Biofilms are microbial cells that are protected by a self-generated matrix of extracellular polymeric substances. In addition to their intrinsic antibiotic resistance, these biofilms are able to respond to the stresses from the antibiotic by inducing drug resistance mechanisms. Currently, the strategy to combat drug resistance is to develop novel drugs, however, the rate of drug development is being surpassed by the rate of drug resistance. There is therefore a need for alternative means in enhancing the efficacy of current drug therapeutics. We propose to use of high intensity focused ultrasound (HIFU) to disrupt the biofilm and promote drug penetration. However, the effects of HIFU on these bacterial communities remain unknown. Here we report on microstructural changes within biofilms formed by *Pseudomonas aeruginosa* due to exposure to HIFU at 500 kHz center frequency. Changes to the biofilm were nondestructively measured through impedance spectroscopy and confocal microscopy. Biofilms were shown to induce cavitation (as measured by a passive cavitation detector) at relatively low pressure amplitudes suggesting the presence of cavitation nuclei within the extracellular matrix.

2:15

4pBAa4. Clot stiffness is inversely correlated with rt-PA thrombolytic efficacy *in vitro*. Karla P. Mercado-Shekhar (Internal Medicine, Univ. of Cincinnati, 231 Albert Sabin Way, Cardiovascular Ctr. 3944, Cincinnati, OH 45267-0586, karlapatricia.mercado@uc.edu), Robert Kleven (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Hermes Aponte Rivera, Ryden Lewis, Kunal B. Karani (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH), Hendrik J. Vos (Biomedical Eng., Erasmus Medical Ctr., Rotterdam, Netherlands), Todd A. Abruzzo (Neurosurgery, Univ. of Cincinnati, Cincinnati, OH), Kevin J. Haworth, and Christy K. Holland (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Predicting thrombolytic susceptibility to recombinant tissue plasminogen activator (rt-PA) *a priori* could help guide clinical decision-making during acute ischemic stroke treatment and avoid adverse off-target lytic effects. The composition and structure of clots impact their mechanical properties and rt-PA thrombolytic efficacy. The goal of this study was to determine the relationship between clot elasticity and rt-PA thrombolytic efficacy *in vitro*. Human and porcine retracted and unretracted clots were fabricated in glass pipettes. Clots were embedded in agar phantoms, and their Young's moduli were estimated using single-track-location shear wave elasticity imaging. The rt-PA thrombolytic efficacy was evaluated *in vitro* using the percent clot mass loss. The Young's moduli of unretracted porcine and human clots (1.68 ± 0.18 kPa and 0.72 ± 0.13 kPa, respectively) were significantly lower ($p<0.05$) than those of retracted porcine and human clots (4.96 ± 1.07 kPa and 3.38 ± 1.82 kPa, respectively). The percent mass loss of unretracted porcine and human clots ($28.9\pm6.1\%$ and $45.2\pm7.1\%$, respectively) were significantly higher ($p<0.05$) than those of retracted porcine and human clots ($10.9\pm2.1\%$ and $25.5\pm10.0\%$, respectively). The results revealed a linear inverse correlation between the Young's moduli and percent clot mass loss ($R^2=0.95$, $p=0.025$), suggesting that clot stiffness may serve as a surrogate metric for rt-PA thrombolytic susceptibility.

2:30

4pBAa5. *In situ* hydrogel formation for biomedical applications using acoustic cavitation from high intensity focused ultrasound. Umesh Jonnalagadda (School of Chemical and Biomedical Eng., Nanyang Technolog. Univ., Singapore, Singapore), Feifei Li (Div. of Chemistry and Biological Chemistry, Nanyang Technolog. Univ., Singapore, Singapore), Jim Lee (Inst. of Chemical and Eng. Sci., Agency for Sci., Technol. and Res., Singapore, Singapore), Atsushi Goto (Div. of Chemistry and Biological Chemistry, Nanyang Technolog. Univ., Singapore, Singapore), Minh Nguyen (Inst. of Chemical and Eng. Sci., Agency for Sci., Technol. and Res., 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)

There is a growing interest in polymer mechanochemistry for their industrial applications. For example, stress-induced crosslinking gel formation from polymer networks is a rapidly growing field of study. Recent work utilizes a variety of different polymer structures and crosslinking mechanisms. However, these polymers are typically soluble in only organic solvents and require the use of a sonicating probe or bath at frequencies below 100 kHz. These requirements limit their use in biomedical applications that require *in situ* gel formation within a patient (e.g., blocking of varicose veins, internal wound healing, etc.). Here we report on the development of a water soluble block copolymer that forms a hydrogel in the presence of acoustic cavitation from high intensity focused ultrasound. These block copolymers are comprised of hydrophilic polyethylene glycol methyl methacrylate units and hydrophobic tridentate crosslinkers. The tridentate crosslinker forms bonds with free metal ions in solution only in the presence of acoustic cavitation induced mechanical stress. We show that the block copolymer is capable of forming a hydrogel in under 90 seconds and will also block a liquid channel formed in an agarose cylinder.

2:45

4pBAa6. Blood coagulation monitoring using acoustic levitation. Vahideh Ansari (Mech. Eng., Boston Univ., 110 Cummington Mall, Rm 414, Boston, MA 02215, vansari@bu.edu), Carol Brugnara (Dept. of Lab. Medicine, Boston Children's Hospital, Boston, MA), and R. G. Holt (Mech. Eng., Boston Univ., Boston, MA)

Impaired blood coagulation can result from a variety of conditions including severe trauma, illness or surgery, and can cause life-threatening bleeding or thrombotic disorders. As a result, instruments which measure functional changes in blood properties upon activation of the clotting cascade are crucial to assess coagulopathies in various disease states. Thromboelastography (TEG) is the gold standard for whole blood coagulation monitoring. TEG requires the blood to be in contact with the sample holder and sensor, leading to a variety of potential artifacts in the results. Acoustic levitation provides non-contact containment and manipulation. We have levitated microliter drops of blood, and employed a drop eigenmodal oscillation technique in order to infer viscoelastic material properties of blood as it coagulates. By comparing our results with TEG results on the same samples, we are able to illustrate certain advantages of the levitation technique. [Work supported by NSF grant # 1438569.]

3:00

4pBAa7. Disposable device for human plateletpheresis. Yuyang Gu, Chuyi Chen, Zeyu Wang, Po-Hsun Huang (Mech. Eng. and Mater. Sci., Duke Univ., B342 Levine Sci. Res., Duke University, Durham, NC 27708, yg90@duke.edu), Hai Fu (Dept. of Fluid Control and Automation, School of Mechatronics Eng., Harbin Inst. of Technol., Durham, North Carolina), and Tony Jun Huang (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC)

Plateletpheresis is a crucial step for the blood donation and clinical diagnosis. Herein we demonstrate a method to conduct plateletpheresis using

acoustofluidic technology in a disposable device that is fabricated solely by a plastic material with precise simulated modeling using low-cost technic. In this device, a quarter wavelength resonator is utilized in order to serve the top layer as the 'tracking wall.' With this device, over 87% RBC removal and platelet recovery rate are achieved at 20 mL/min flow rate with undiluted whole human blood while the function of the platelets is still retained and better than conventional plateletpheresis method. Conclusively, this disposable device synthesizes the high efficiency, high throughput and high biocompatibility which is suitable as the substitute of the current plateletpheresis equipment.

THURSDAY AFTERNOON, 10 MAY 2018 GREENWAY F/G, 3:30 P.M. TO 5:30 P.M.

Session 4pBAb

Biomedical Acoustics: Imaging

Jeffrey A. Ketterling, Chair
Riverside Research, 156 William St, New York, NY 10038

Contributed Papers

3:30

4pBAb1. Simulation and analysis of three-dimensional echo decorrelation imaging. Michael T. Cox, Mohamed A. Abbass, and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH 45267-0586, cox2mt@mail.uc.edu)

A numerical model for pulse-echo ultrasound imaging [Mast, JASA 128:EL99-EL104, 2010] is extended to three-dimensional (3D) imaging of linear, weakly scattering continuum media by matrix array transducers. In this model, beam patterns for steered transmit and receive subapertures, calculated analytically under the Fresnel approximation, are combined with a 3D numerical tissue model to yield beamformed scan lines in a rectilinear or pyramidal configuration. Simulated scan lines are processed to yield both volumetric B-mode images and echo decorrelation images, comprising spatial maps of the normalized decorrelation between sequential pulse-echo image volumes. A method is demonstrated for construction and scan conversion of quantitative 3D echo decorrelation images such that, as for 2D echo decorrelation imaging [Hooi *et al.*, JASA 137:585-597, 2015], the mapped echo decorrelation consistently estimates the local decoherence spectrum of the tissue reflectivity. Simulated 3D echo decorrelation images of random media are shown to accurately estimate known decoherence distributions that simulate heating effects in thermal ablation. Effects of factors including transducer frequency, scan configuration, and simulated lesion size on accuracy of decoherence estimates are tested. Applications to 3D monitoring of radiofrequency and microwave ablation are discussed.

3:45

4pBAb2. Exploiting Ballou's rule for better tissue classification. Johannes Kvam, Bjørn A. J. Angelsen (Dept. of Circulation and Medical Imaging, Norwegian Univ. of Sci. and Technol., Det medisinske fakultet, Institutt for sirkulasjon og bildediagnostikk, Postboks 8905, Trondheim, Sør-Trøndelag 7491, Norway, johannes.kvam@gmail.com), and Sverre Holm (Dept. of Informatics, Univ. in Oslo, Oslo, Norway)

Tissue characterisation using the parameter of nonlinearity B/A has shown promising diagnostic value in medical imaging. Its estimation usually infers an assumption of constant sound speed and density. However, an empirical result known as Ballou's rule shows a significant correlation between the reciprocal sound speed and the parameter of nonlinearity for liquid metals. Since the 60's, this result has been found also to be applicable for soft tissues as well with a high correlation ($r = 0.83$). The correlation between B/A and compressibility, the inverse bulk modulus, is just as strong ($r \geq 0.83$) indicating that the product of the two can be beneficial for two main reasons. First, as the variables are highly correlated the variation in the product is larger than that of B/A by itself. This allows for a higher sensitivity as the separation between tissue types will be larger. Second, as methods for estimating B/A often assume constant sound speed and density, variations in these parameters result in uncertainty in the estimated B/A values. By estimating the product of B/A and compressibility instead, the accuracy of the estimate improves. There are multiple methods for estimating this product including the dual frequency SURF method (Hansen *et al.*, JASA 2010).

4:00

4pBAb3. High-frequency (10-100 MHz) ultrasound instrumented forceps for precision cancer surgery. Timothy E. Doyle (Phys., Utah Valley Univ., MS 179, 800 W. University Parkway, Orem, UT 84058-5999, Timothy.Doyle@uvu.edu), Michael J. Salisbury (Manufacturing Eng., Utah Valley Univ., Orem, UT), Michael J. Bennett (Phys., Utah Valley Univ., Orem, UT), Garrett Wagner (Comput. Eng., Utah Valley Univ., Orem, UT), Caitlin Carter (BioTechnol., Utah Valley Univ., Draper, UT), Tobee P. Gunter, Jessica E. Carlson (Biology, Utah Valley Univ., Orem, UT), Nicole Cowan (BioTechnol., Utah Valley Univ., Orem, UT), Johnathon S. Gay (Phys., Utah Valley Univ., Orem, UT), and Cindy B. Matsen (Surgery, Univ. of Utah, Salt Lake City, UT)

A critical issue for surgery of soft tissue cancers is ensuring removal of all malignant tissue while conserving as much unaffected tissue as possible. A minimally invasive, *in vivo* technique for detecting residual cancer in surgical margins or metastatic cancer in lymph nodes would significantly improve the precision and efficacy of cancer surgery. The objective of this project was to develop ultrasound instrumented "smart" forceps to guide surgeons during operations, to provide instant diagnostic information, and to enable more precise and complete resection of malignant tissue. The device was based on Martin forceps, modified to include two ultrasonic sensors (polyvinylidene difluoride) at the tips of the forceps, a tissue thickness sensor (rotary potentiometer), sensor wires, and a spring to control the manual stiffness of the forceps. A high-frequency pulser-receiver was used to excite the transmitting sensor at 50 MHz and amplify through-transmission signals from the receiving sensor. The forceps were tested with agarose phantoms embedded with polyethylene microspheres of different diameters (58-550 μm) to vary attenuation. Testing included standard 50-MHz immersion transducers for comparison. The results showed that the forceps' performance was equal to standard transducers, with comparable sensitivity for attenuation and superior accuracy for wave speed.

4:15

4pBAb4. Gas-generating nanoparticles for molecular ultrasound imaging. In-Cheol Sun and Stanislav Emelianov (ECE and BME, Georgia Inst. of Technol., 777 Atlanti Dr., Atlanta, GA 30332-0250, stas@gatech.edu)

Diagnostic ultrasound imaging often relies on contrast agents. Most common contrast agents are microbubbles that are confined to vascular compartments and molecular targets because micrometer size particles cannot escape through endothelial barriers. Therefore, a desired ultrasound contrast agent should consist of nanometer scale particles that are capable of escaping from vasculature, penetrating into tissue, and then generating sufficient contrast once they reach the target site. We developed a novel contrast agent—plasmonic nanoparticles covered by azide compounds (AzNPs), capable of on-demand laser-induced gas generation via photolysis of azide functional groups. The 50 nm diameter AzNPs were small enough to penetrate the endothelial barrier. Upon laser activation, the AzNPs generated nitrogen gas which served as ultrasound contrast agent. The feasibility of the AzNPs to produce ultrasound contrast was tested within polyethylene tube under laminar flow. Upon laser irradiation, significant ultrasound signal enhancement was observed due to the N2 gas bubbles generated by the photolysis of AzNPs. These and other studies suggest that the AzNPs may enable the ultrasound diagnosis of various diseases that conventional microbubbles cannot detect. Furthermore, these optically absorbing particles can also be used for imaging and therapeutic applications of light and sound such as ultrasound-guided photoacoustic imaging.

4:30

4pBAb5. Plane-wave vector-flow imaging of adult mouse heart. Jeffrey A. Ketterling (F.L. Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St, New York, NY 10038, jketterling@riversideresearch.org), Akshay Shekhar, Orlando Aristizabal (Skirball Inst. of Biomolecular Medicine and the Dept. of Radiology, New York Univ. School of Medicine, New York, NY), Anthony Podkowa (Elec. and Comput. Eng., Univ. of Illinois, Urbana, IL), Billy Y. Yiu, and Alfred Yu (Dept. of Elec. and Comput. Eng., Univ. of Waterloo, Waterloo, ON, Canada)

High-frequency ultrasound Doppler modes have been used extensively for murine cardiovascular (CV) studies, but traditional linear-array imaging modes are limited in terms of spatial and temporal resolution. Plane-wave imaging methods allow for high-speed vector-flow information to be obtained throughout a full image frame. Plane-wave imaging has been demonstrated in human CV studies, but its use in mouse models has received minimal attention. A Verasonics Vantage with an 18-MHz linear array was used to acquire plane-wave data at a frame rate of 30 kHz from the left ventricle of adult mice. Batches of 3 transmissions spanning $\pm$ 5 degrees were sent out. The mouse was placed supine on a heated imaging platform and then 2D + time data sequences. The data were beamformed using standard delay-and-sum methods and vector-flow estimates were obtained at each pixel location using a least-squares, multi-angle Doppler analysis approach. Vortex patterns in the left ventricle were visualized over several heart cycles showing the complex flow in the mouse heart.

4:45

4pBAb6. Using mean frequency estimation algorithms for unambiguous identification and visualization of an acoustically active catheter. Viksit Kumar, Bae Hyeong Kim (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine and Sci., 200 First St. SW, Rochester, MN 55905, kumar.viksit@mayo.edu), Azra Alizad (Radiology, Mayo Clinic College of Medicine and Sci., Rochester, MN), Marek Belohlavek (Cardiology, Mayo Clinic College of Medicine and Sci., Scottsdale, AZ), and Mostafa Fatemi (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine and Sci., Rochester, MN)

Minimally invasive procedures using ultrasound are becoming prominent, especially in applications like biopsy, catheter guidance, and surgeries. In our previous work a catheter tip is fitted with a miniature omnidirectional piezoelectric crystal; the ultrasonic field from the imaging array and the piezoelectric crystal interact resulting in a symmetric Doppler shift. Previously, a frequency domain algorithm was used to identify pixels having the symmetric Doppler shift. The identified pixels can then be visualized on B-mode and color map with a unique color thus representing the location of the catheter tip. However, frequency domain algorithms are computationally expensive and require changes to data workflow. Using mean frequency estimators with symmetric Doppler shift results in the catheter tip being displayed as zero mean noise making it hard to distinguish from color Doppler noise. To overcome this limitation, complex demodulation can be used. Modulating the time domain signal with sine and cosine by the piezoelectric excitation frequency and summing them up results in cancellation of one of the symmetric Doppler components. With only one frequency component remaining mean frequency estimators can be used to identify the location of the catheter. The presented algorithm has lower computational complexity and requires minimal change in software design.

5:00

4pBAb7. Detection of lymph node metastasis using photoacoustic imaging and glycol-chitosan-coated gold nanoparticles. Diego Dumani, In-Cheol Sun, and Stanislav Emelianov (School of Elec. and Comput. Eng. and Dept. of Biomedical Eng., Georgia Inst. of Technol. and Emory Univ. School of Medicine, 777 Atlanti Dr., Atlanta, GA 30332-0250, stas@gatech.edu)

Metastases, rather than primary tumors, are responsible for majority of deaths in cancer patients. We developed a non-invasive method to detect sentinel lymph node (SLN) metastasis using combined ultrasound and photoacoustic (US/PA) imaging with glycol-chitosan-coated gold nanoparticles (GC-AuNP). The spatio-temporal distribution of GC-AuNP is affected by presence of metastasis, which is detected by US/PA. GC-AuNPs (100 μl, 0.1 mg Au/ml) were injected peritumorally in breast tumor-bearing mice, and allowed to drain for 24 hours into the SLN. The SLN was located using b-mode ultrasound, and multi-wavelength PA was used to detect GC-AuNPs. Spectroscopic analysis allowed to isolate the signal from GC-AuNPs and quantify their accumulation in the SLN after cellular uptake and transport by immune cells. Results show that distribution of GC-AuNPs in the SLN was affected by presence of metastasis. The overall accumulation was also reduced, with more than 20% signal decrease compared to non-metastatic controls. Histological analysis confirmed that distribution of GC-AuNP-containing immune cells is reduced due to presence of metastatic cells. Our method successfully distinguishes metastatic from non-metastatic lymph nodes using biocompatible nanoparticles and could aid physicians in detection of micrometastasis thus guiding and avoiding unnecessary SLN biopsy.

5:15

4pBAb8. Modeling element directivity in minimum variance beamforming for medical ultrasound. Brian H. Tracey (ECE, Tufts Univ., 196 Boston Ave., Medford, MA 02155, btracey@eecs.tufts.edu), David Lemmerhirt (Sonetics, Inc., Ann Arbor, MI), Dominique Penninck (Clinical Sci., Tufts Cummings School of Veterinary Medicine, Grafton, MA), and Joseph F. Polak (Radiology, Tufts School of Medicine, Boston, MA)

Minimum variance (MV) adaptive beamforming, which adaptively estimates and suppresses interfering signals, has attracted increased attention for ultrasound imaging in recent years. While many MV algorithms have been applied to ultrasound, these algorithms generally rely on signal models that neglect amplitude differences across the array due to element directivity or other factors, introducing mismatch for near-array sources. This paper proposes a beamspace method using randomly sampled subarrays which allows the use of higher-fidelity signal models. Using experimental data, we demonstrate that this algorithm yields noticeable gains in signal contrast by accounting for element directivity.

Session 4pMU

Musical Acoustics, Psychological and Physiological Acoustics, and Signal Processing in Acoustics: Sound Effects and Perception

Jonas Braasch, Chair
School of Architecture, Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180

Chair's Introduction—1:00

Invited Papers

1:05

4pMU1. Travelling by microphone—The origami of time- and level-based effects. M. Torben Pastore (Speech and Hearing Sci., Arizona State Univ., 4 Irving Pl., Troy, New York 12180, m.torben.pastore@gmail.com)

The introduction of the microphone brought radical transformations to vocal technique—consider the transition from Bel Canto to crooning. This radically changed the relation between the singer and the backing band or orchestra. More importantly, the singer could now appeal intimately to the listener with all the directness and art of a whispering lover, but with the full backing of the band for sympathy, illustration, and reverie. With the introduction of mutitrack recording and sound reinforcement, artists could bring their listeners closer, even behind their eyes, or hold a space for listeners to relate to themselves, by presenting multiple spaces with multiple actors at multiple distances simultaneously. The simple use of microphones, delays, reverbs (modulated and not), compression, and other effects to say what cannot be said with time and space-bound forms where the relations between musicians, singers and listeners are defined and constrained by the single space they share will be considered. The manifestation of the psychedelic/deep listening experience as a transform, via simple time- and level-based effects, of music into experience grounded in actual, unknowable reality will be considered and auditioned.

1:25

4pMU2. Repurposing the algorithm: How digital signal processing brought on the re-use of reverb. Nikhil Deshpande and Jonas Braasch (Architecture, Rensselaer Polytechnic Inst., 220 3rd St., Troy, NY 12180, deshpn@rpi.edu)

The earliest digital reverberation algorithms were designed to provide sound with a wider and more immersive sense of physical time-based space. These algorithms successfully replicated the perceptual attributes of realistic rooms, but there was also an interesting and perhaps unforeseen secondary outcome. The ubiquity of digital reverberation made the creative misuse of these algorithms to construct physically impossible spaces readily accessible to musicians. By providing users with access to algorithm controls, reverb grew from a tool for constructing realistic perceptions of rooms into the creation of sound design—and often, the centerpiece of composition. Rather than provide a perceptually accurate presentation of sounds in a stereo field, artists instead began to approach reverb as a smaller part of larger timbral systems. Examples span from reverse reverb to the more complex Eno-based shimmer system, and reverb's place in production and perception will be considered and discussed.

1:45

4pMU3. Audio-haptic perception in immersive improvisational environments. Doug Van Nort (York Univ., 4700 Keele St, Toronto, ON M3J 1P3, Canada, dvnt.sea@gmail.com)

This presentation describes a recent study that explored audio-haptic perception in group performance contexts. Vocalizing and listening activities were structured using methods from the Deep Listening practice developed by Pauline Oliveros, with augmentations to the immersive environment made using a multi-channel audio system, in combination with under-floor and body-worn vibrotactile transducers. Bio-signals from participants were captured and qualitative feedback solicited in order to understand the perceptual and physiological effects of introducing signals to these structured performative activities that are dynamically spatialized in audio as well as haptic sense modalities. The study compares the differences in response when these signals are driven by an external source in comparison to being driven by the bio-signals of participants, including measures focused on convergence and divergence of bio-phyiscal activity. The presentation will compare these scenarios and discuss the larger implications for developing immersive interactive environments that contribute to a sense of connection, sociality and play.

2:05

4pMU4. Guitar amplifier modeling—Perceptual evaluation of audio similarity. Felix Eichas and Udo Zoelzer (Elec. Eng., Helmut Schmidt Univ., Holstenhofweg 85, Hamburg 22043, Germany, udo.zoelzer@hsu-hamburg.de)

Virtual analog modeling of guitar equipment, especially tube-based guitar amplifiers, is a hot topic in sound effect research. However, there are no established standards on the evaluation of the accuracy of such modeling processes. This work presents a first approach on finding a metric for evaluating the similarity between the output of an analog reference amplifier and a digitized version of the same device. A gray-box modeling procedure is used where only assumptions about the general structure of a guitar amplifier are made. The necessary information about the reference device is obtained by input and output measurements. First a digital model is constructed and the error between digital model and analog system is minimized by the Levenberg-Marquardt parameter optimization algorithm. The error is expressed as the result of a cost function. To be able to produce good results, this cost function needs to consider psycho-acoustic aspects like e.g. the perceived frequency resolution. The results of a spectrogram based cost function are compared to the results of a listening test, where the test subjects should rate the similarity of the two audio signals (reference and digital simulation) and the findings are discussed.

2:25

4pMU5. The use of physical and artificial room reverberation to create transparent or fused ensemble sounds. Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu)

Room Reverberation has shaped the sound of musical instruments from the beginning. The oldest known musical instrument, a 40,000-year-old bone flute has been found in a cave, the Hohle-Fels Hoehle. Impulse response measurements taken at this cave reveal that the cave had a reverberation time, T_{30}, at mid frequencies of 1.8 seconds, a value within the range of that of a modern concert hall. Room reverberation has continued to play an important role in defining our cultural sonic environments. During the Reformation, for example, there was a clear call for transparency so that the words of the reverends could be understood by the people. This sound ideal also affected the music of protestant composers like Johann Sebastian Bach an ideal that culminated in the interpretations of Glenn Gould, who preferred the natural sound of the dry sound recording studio. While a concert hall is always bound to its sound, most modern recording studios offer a flexible design where the room can be created artificially around the artist to achieve a certain sound ideal. Unlike in a physical hall that embeds all musicians in a homogenous sound, artificial reverb is often used to provide a unique environment for each instrument to meet specific goals.

2:45

4pMU6. Correlation between perception and acoustical parameters in five musical genres. Rolf Bader (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, R_Bader@t-online.de)

To investigate the influence of sound effects on the perception of musical genres, excerpts from 150 musical pieces of Rock, Jazz, Classical, Ethnographic and Electronic music were compared. In a listening test subjects were asked to judge them in terms of 12 adjectives related to spaciousness, timbre and semantic attributes. Additionally the pieces were analyzed with three methods. To judge the spaciousness the Interaural Cross Correlation (IACC) was calculated. To estimate the audio effect strength an echodensity was defined as the amount of phase fluctuations. To estimate the density of tonal texture a fractal correlation dimension was used. The musical genres clearly sorted along their IACC values with Rock being most 'mono' and Classical music most spatial. The echodensity was about the same for all genres except for electronic music where it was considerably larger. The mean fractal correlation dimensions were about four with all genres pointing to a mean textual density for all genres. In terms of perception, the higher echodensity correlated positively with the adjective "artificial" for Electronic music. Additional findings were made.

3:05–3:20 Break

3:20

4pMU7. Sound effects without side effects—Suppression of artifacts in audio signal processing. Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St, Ste. 3, Lowell, MA 01854, alex@fermata.biz)

Sound effects processes intending one production outcome can too often be accompanied by unwanted side effects. Signal processing strategies have evolved in popular music production that preserve a producer's sonic goals while suppressing these artifacts. The solutions found can be surprising and frequently have no analog in all-acoustic music creation and presentation. Sound effects with these built-in defense mechanisms become studio-only creations, effects for loudspeaker mediated art only.

3:40

4pMU8. Preference, localization, attention, and the limit of localization distance (LLD). David H. Griesinger (Res., David Griesinger Acoust., 221 Mt Auburn St. #504, Cambridge, MA 02138, dgriesinger@verizon.net)

Recent work by Lokki *et al.* has found that a perception he calls "proximity" is a major component of preference in concert halls. In this talk we propose that proximity is perceived when sources are close enough or clear enough that they can be sharply localized and understood. In halls and rooms we find this ability abruptly disappears at a critical distance from the source, the Limit of Localization Distance, or LLD. We find that proximity attracts and holds attention, which is a major reason cinemas and drama theaters have dry acoustics and strong direct sound paths. In this talk we will describe the physics behind the perception of proximity, why it has not been previously recognized, how to predict where the LLD will occur from a binaural impulse response, and how to optimize proximity in halls, theaters, and classrooms.

4:00

4pMU9. Effects created by time-variant spectral processing of sounds. James W. Beauchamp (Elec. & Comput. Eng. and School of Music, Univ. of Illinois at Urbana-Champaign, 1002 Eliot Dr, Urbana, IL 61801-6824, jwbeauch@illinois.edu)

Various processes can be used to modify a sound to produce special effects. One approach is to first compute a time-frequency representation of a sound signal, follow that by altering the representation in various ways, and finally perform resynthesis to produce modified versions of the signal. Example modifications are (1) duration modification (time-scaling), (2) pitch change, (3) noise content manipulation, (4) spectral envelope modification, and (5) temporal envelope modification. The time-frequency representation can also be used to develop simplified synthesis models that depend on only a few time-varying parameters, such as overall amplitude, pitch, and spectral centroid and shape. In the special case of musical sounds with vibrato, spectral amplitudes and frequencies can be parameterized in terms of time-variant vibrato depth, rate, and mean envelope. These parameters can then be controlled independently of pitch and duration. Also, timbral parameters of instruments can be interchanged or blended to create hybrid instruments with novel timbral qualities. Several of these effects will be described and demonstrated.

4:20

4pMU10. Allpass decorrelating filter design and evaluation. Elliot K. Canfield-Dafilou and Jonathan S. Abel (Ctr. for Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., 660 Lomita Dr, Stanford, CA 94305, kermit@ccrma.stanford.edu)

By breaking up the phase coherence of a signal broadcast from multiple loudspeakers, it is possible to control the perceived spatial extent and location of a sound source. This so-called signal decorrelation process is commonly achieved using a set of linear filters, and finds applications in audio upmixing, spatialization, and auralization. It is important that any individual decorrelating filter does not perceivably alter the sound of its input, i.e., that it does not color the signal spectrum or smear transients over time. In that way, the process will only affect the spatial properties of the sound. Allpass filters make ideal decorrelation filters since they have unit magnitude spectra, and therefore can be perceptually transparent. By manipulating their phase, allpass filters can be leveraged to achieve a degree of decorrelation. Here, we present a method for designing allpass decorrelation filters by specifying group delay trajectories in a way that allows for control of the amount of correlation as a function of frequency. This design is efficiently implemented as a cascade of biquad allpass filters. We present statistical and perceptual methods for evaluating the amount of decorrelation and audible distortion.

Contributed Papers

4:40

4pMU11. Pitch and timbre discrimination at wave-to-spike transition in the cochlea. Rolf Bader (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, R_Bader@t-online.de)

A new definition of musical pitch is proposed. A Finite-Difference Time Domain (FDTM) model of the cochlea is used to calculate spike trains caused by tone complexes and by a recorded classical guitar tone. All harmonic tone complexes, musical notes, show a narrow-band Interspike Interval (ISI) pattern at the respective fundamental frequency of the tone complex. Still this fundamental frequency is not only present at the bark band holding the respective best frequency of this fundamental frequency, but rather at all bark bands driven by the tone complex partials. This is caused by drop-outs in the basically regular, periodic spike train in the respective bands. These drop-outs are caused by the energy distribution in the wave form, where time spans of low energy are not able to drive spikes. The presence of the fundamental periodicity in all bark bands can be interpreted as pitch. Contrary to pitch, timbre is represented as a wide distribution of different ISIs over bark bands. The definition of pitch is shown to also works with residue pitches. The spike drop-outs in times of low energy of the wave form also cause undertones, integer multiple subdivisions in periodicity, but in no case overtones can appear. This might explain the musical minor scale, which was proposed to be caused by undertones already in 1880 by Hugo Riemann, still until now without knowledge about any physical realization of such undertones.

4:55

4pMU12. The time and place of cochlear implant pitch perception. Susan R. Bissmeyer (Biomedical Eng., Univ. of Southern California, 1042 Downey Way DRB 140, Los Angeles, CA 90089-1111, ssubrahm@usc.edu), Shaikat Hossain, and Raymond L. Goldsworthy (Otolaryngol., Univ. of Southern California, Los Angeles, CA)

It is well established that cochlear implant users can perceive pitch associated with pulse train stimulation rate, but that rate sensitivity diminishes above 300 pps. However, there have been multiple reported cases of individuals sensitive to stimulation rates well above 300 pps. Furthermore, Goldsworthy and Shannon (2014) demonstrated that sensitivity to stimulation rate could be improved through psychophysical training. The goal of the present study is to identify factors affecting rate sensitivity in cochlear implant users, including age, etiology, and duration of deafness, electrode location, stimulation mode, neural health as measured by multipulse integration, and spatial tuning as measured by forward masked thresholds. Adult cochlear implant users serve as subjects in a 3-week protocol that includes 4 hours per week of electrode psychophysics conducted in the laboratory and 30 minute daily auditory listening exercises conducted at home. Results indicate that cochlear implant users with minimal psychophysical training consistently discriminate rate differences at least as high as 800 pps. Emerging trends show a group effect with a monopolar stimulation mode and basal cochlear location benefit. We will report on the effect of the factors affecting rate sensitivity for this ongoing study.

5:10

4pMU13. Examining the auditory cortex response to musical stimuli with differences in timbre and reverberation. Martin S. Lawless and Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, msl224@psu.edu)

In a previous functional magnetic resonance imaging (fMRI) study by the authors, the addition of reverberation was shown to degrade the cortical auditory response to musical stimuli. This degradation may occur since reverberation blurs the frequency content of the stimuli in time, making it more difficult for the brain to distinguish between distinct auditory objects. The current study aims to further establish the auditory response to reverberant stimuli by investigating if higher-order frequency content (timbre) affects the auditory cortex's sensitivity to reverberation. Using an anechoic solo-instrumental trumpet motif, a sine-trumpet musical stimulus was generated by removing the harmonic content from the played notes while maintaining the attack of the instrument and rhythm of the musical passage. Auralizations of both motifs were created from eight simulated room conditions with reverberation times ranging from 0.0-7.2 s. Participants were recruited to rate the auralizations in terms of overall preference, perceived reverberance, and perceived clarity. Subsequently, the participants listened to a subset of the solo-instrumental and sine-trumpet auralizations in an MRI machine. Contrasts between the motifs and room acoustic conditions were analyzed to confirm the sensitivity of the auditory cortex to reverberation, as well as investigate the possible interaction effect between reverberation and timbre.

Session 4pNS

Noise, Psychological and Physiological Acoustics, and ASA Committee on Standards: Hearing Protection: Impulse Peak Insertion Loss, Specialized Hearing Protection Devices II

Cameron J. Fackler, 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

Invited Papers

1:30

4pNS1. Observation on the variability of fitting by two experimenters at REAT and the agreement with objective measurement on three premolded earplugs. Yufei Liu (Person Safety Div., 3M, No.9, Nanxiang Er Rd., Sci. City, Guangzhou, Guangdong 510663, China, sliu9@mmm.com)

Real ear attenuation at threshold (REAT) has been recognized as the gold standard method to evaluate the attenuation of hearing protection devices (HPDs). It has been standardized in ANSI, ISO/EN and AS/NZS documents. The fitting of HPDs is one of the key factors of REAT measurement. The experimenter fit protocol of ANSI S3.19 is considered one precise approach to obtain the "best fit" of HPDs. A study of the attenuation obtained between the three fits for two experimenters on the same ten subjects on three premolded three-flange earplugs was conducted at 3M E-A-RCAL laboratory, including two models of Chinese brands earplugs and 3M^{TM} E-A-R^{TM} UltraFit earplug. Good agreement was shown between the REAT results that were obtained by the two experimenters. The REAT result of one Chinese brand earplug shows much higher variability within subjects. Comparing with the insertion loss (IL) measured by G.R.A.S 45CB, which has been corrected for bone conduction, occlusion effects and physiological noise, produced IL for two of the tested earplugs that did not showed good agreement with REAT results at most, but not frequencies. The discrepancies will be discussed.

1:50

4pNS2. Quantifying the insertion loss of hearing protection using a compressed gas shock tube. Theodore F. Argo (Appl. Res. Assoc., Inc., 7921 Shaffer Parkway, Littleton, CO 80127, targo@ara.com), Nate Greene (Dept. of Otolaryngol., Univ. of Colorado Anschutz Medical Campus, Aurora, CO), James Easter (Cochlear Boulder LLC, Boulder, CO), Daniel J. Tollin (Dept. of Otolaryngol. and Dept. of Physiol. and Biophys., Univ. of Colorado Anschutz Medical Campus, Aurora, CO), and Timothy J. Walilko (Appl. Res. Assoc., Inc., Littleton, CO)

Compressed gas shock tubes provide a method of generating repeatable and consistent blast overpressure waves similar to those observed in combat and military training. Shock tubes of different diameters and lengths have been used to evaluate the level dependence of various types of hearing protection devices (HPDs) for sound pressure levels ranging from 132 to 192 dB. Insertion loss data was generated using mechanical test fixtures, post mortem human surrogates, and animal models in order to compare the respective auditory responses at these high pressures. The inclusion of animal models was possible since gas driven shock tubes can be housed in mobile laboratories and trailered to vivarium for physiological and behavioral observations. Measurements were conducted on a variety of HPDs and indicate that the responses of many advanced hearing protection devices are nonlinear at these sound pressure levels. Models of hearing protection and auditory injury have been updated and improved based upon quantification of HPD level-dependence. [Work supported by DOD grant W81XWH-15-2-0002.]

2:10

4pNS3. Occlusion effects of hearing protection device on ear canal transfer function and cochlear injury from impulse noise exposure. Brissi Zagadou (L3 Technologies, 10180 Barnes Canyon Rd, San Diego, CA 92121, Brissi.Zagadou@L3T.com)

The objective of this paper is to present a new finding on the effects of hearing protection devices (HPDs) on the ear canal (EC) transfer function (TF_{EC2ED}). Specifically, we show that when an earmuff is worn, the TF_{EC2ED} is not the same as that for the bare head, leading to a significant difference in cochlear injury prediction. We used shock tube blast with an acoustical test fixture (ATF) fitted with the recovered earmuffs from the historic blast overpressure project (BOP) to measure the TF_{EC2ED}. A microphone was placed at the

EC entrance under the earmuff (undermuff) to measure the pressure at the EC entrance, while the built-in ATF's microphone was used to measure the eardrum (ED) pressure. The TF_{EC2ED} is defined as the ratio of the pressure at the ED to the pressure at the EC entrance. We compared the cochlear integrated energy (ICE) values, when the TF_{EC2ED} obtained from bare head (Case-1) and that measured using the ATF fitted with an earmuff (Case-2), respectively are used in the ICE-model, to estimate the effects of the difference between the TF_{EC2ED}s on injury. The human BOP undermuff pressure data were used as model input. For Case-1, the input was at the EC. The ED pressure, reconstructed using the undermuff pressure and the TF_{EC2ED} measured using the ATF, was used for Case-2. We found significant differences in the ICE-values as the result of the difference between the TF_{EC2ED}s. We conclude that hearing protector models must correctly model the occlusion effects of HPDs by incorporating the change in the TF_{EC2ED}.

2:30

4pNS4. Anticipatory middle ear muscle contractions in damage-risk criteria. Gregory Flamme (SASRAC, 2264 Heather Way, Forest Grove, OR 97116, gflamme@sasrac.com), Stephen M. Tasko (SASRAC, Kalamazoo, MI), Kristy Deiters (SASRAC, Portage, MI), William J. Murphy (National Inst. for Occupational Safety and Health, Cincinnati, OH), Heath Jones, William A. Ahroon (U.S. Army Aeromedical Res. Lab., Fort Rucker, AL), and Nate Greene (U.S. Army Aeromedical Res. Lab., Aurora, CO)

Anticipatory middle ear muscle contractions (MEMC) have been implemented as protective components of Damage-Risk Criteria (DRC) for impulsive noises. However, no studies have shown that anticipatory MEMC are pervasive among humans. This presentation describes a series of studies of the viability of assumed anticipatory MEMC obtained either through classical conditioning or while operating a model gun. Participants were adults with normal hearing, and the conditioning tasks varied on sensory modality and attention. Both between- and within-subjects designs were used. A conditioned response was defined as an MEMC occurring prior to the unconditioned stimulus and when only the conditioning stimulus was presented. These studies do not suggest that anticipatory MEMC should be included in DRC for impulsive noises.

Contributed Papers

2:50

4pNS5. Developing a method to assess noise reduction of firearm suppressors for small-caliber weapons. William J. Murphy (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov), Gregory Flamme (SASRAC, Forest Grove, OR), Adam R. Campbell, Edward L. Zechmann (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, Cincinnati, OH), Michael Stewart (Dept. Commun. Disord., Central Michigan Univ., Mount Pleasant, MI), Stephen M. Tasko (Dept of Speech, Lang. and Hearing Sci., Western Michigan Univ., Kalamazoo, MI), James E. Lankford (Allied Health and Commun. Disord., Northern Illinois Univ., Dekalb, IL), Deanna K. Meinke, and Donald Finan (Audiol. and Speech-Lang. Sci., Univ. of Northern Colorado, Greeley, CO)

Firearm suppressors have the potential to reduce the muzzle blast by diffusing the initial shock wave of a gunshot. Currently, the American National Standards Institute does not have any standards that specifically address suppressor measurements. A recent NATO standard, AEP 4875 has been proposed to characterize suppressor performance, but the scope of this standard does not include suppressor effects at the shooter's ear. Additionally, the standard requires firing the weapon from an elevated platform 4 meters above the ground with microphones positioned with regular spacing of about 18 degrees at 5 meters from the muzzle. This study evaluated fourteen different firearms with and without a suppressor. Different loads of ammunition were used to vary the speed of the projectile. For ten of the guns, both supersonic and subsonic conditions were measured. Twelve microphones were positioned at 30-degree spacing in 3-meter ring at 1.5 meters above the ground. One microphone was positioned at 1 meter to the left of the muzzle and two microphones were positioned at 15 centimeters from the right and left ears. The suppressors were effective in reducing the peak sound pressure levels between 3 and 28 dB and 8-hour equivalent energy (LAeq8) between 2 and 24 dB.

3:05

4pNS6. Impulse noise measurements of M16 rifles at Quantico Marine Base. Reese D. Rasband (Dept. of Phys. and Astronomy, Brigham Young Univ., BYU N283 ESC, Provo, UT 84602, r.rasband18@gmail.com), Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH), Kent L. Gee, S. Hales Swift (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Caleb M. Wagner (Human Systems Program Office, Air Force Res. Lab., Dayton, OH), William J. Murphy, and Chucri A. Kardous (Hearing Loss Prevention Team, National Inst. for Occupational Safety and Health, Cincinnati, OH)

This paper describes a study conducted at U.S. Marine Corps Base Quantico to determine firing range impulse noise levels and assess noise exposures. Measurements were performed with M16 rifles at an outdoor firing range using a 113-channel array of 6.35 and 3.18 mm microphones that spanned potential locations for both shooters and instructors. Data were acquired using 24-bit cards at a sampling rate of 204.8 kHz. Single weapon measurements were made with and without an occupied range, with a shooter and with a remotely triggered gun stand. In addition, measurements were made with multiple shooters to simulate exposures for a realistic range environment. Results are shown for the various range configurations as a function of angle and distance. Analyses include waveforms, spectra, and peak levels, as well as the 100 ms A-weighted equivalent levels required by military standard MIL-STD-1474E. [Sponsored by US Office of Naval Research.]

3:20

4pNS7. Rating the perception of jet noise crackle. Paul B. Russavage, Tracianne B. Neilsen, Kent L. Gee, and S. Hales Swift (Dept. of Phys. & Astronomy, BYU, N283 ESC, Provo, UT 84602, pau616@gmail.com)

Crackle is a perceptual aspect of noise caused by impulsive acoustic shocks and observed in supersonic jets, including those from military aircraft and rockets. Overall and long-term spectral noise metrics do not account for the unique perception of crackle. Listening tests were designed to better understand perception of crackle and examine its relationship to physical noise metrics, such as skewness of the first time derivative of the pressure waveform, hereafter derivative skewness. It is hypothesized that as derivative skewness increases, the perception of crackle tends to increase. Two listening tests were conducted with 30 subjects to examine their perception of crackle. In the first test, subjects compared and ordered crackle-containing sounds. In the second test, category scaling was employed with subjects rating the crackle content with category labels: (1) smooth noise with no crackle, (2) rough noise with no crackle, (3) sporadic crackle, (4)

continuous crackle, and (5) intense crackle. There is a high correlation between perception of crackle and derivative skewness. This five-point classification scheme appears to be an effective tool to measure crackle perception. These insights will help inform community noise models, allowing them to incorporate annoyance due to jet crackle. [Data courtesy of F-35 JPO.]

THURSDAY AFTERNOON, 10 MAY 2018 GREENWAY J, 1:40 P.M. TO 4:15 P.M.

Session 4pPA

Physical Acoustics, Noise and ASA Committee on Standards: Sonic Boom II

Alexandra Loubeau, Cochair
Structural Acoustics Branch, NASA Langley Research Center, MS 463, Hampton, VA 23681

Joel B. Lonzaga, Cochair
Structural Acoustics Branch, NASA Langley Research Center, 2 N. Dryden St., B1208 MS 463, Hampton, VA 23681

Invited Papers

1:40

4pPA1. Sonic boom induced window rattle in indoor environments. Jacob Klos (Structural Acoust. Branch, NASA Langley Res. Ctr., 2 N. Dryden St., MS 463, Hampton, VA 23681, j.klos@nasa.gov)

The perceptual environment inside homes ensonified by sonic booms consists mainly of the transmitted indoor boom sound, rattle from items in loose contact with vibrating structures, and structural vibrations that may be felt or seen. In this presentation, measurements of window rattle from a recent laboratory study conducted inside the Interior Effects Room at NASA Langley Research Center will be summarized. Low amplitude sonic booms were reproduced at the facility exterior, which induced rattle in several distressed windows that were interchanged in the facility wall. The indoor sound field, the combined transmitted boom and window rattle sound, was measured at seven microphones placed inside the test room. The "rattle only" sound was recovered by subtracting the "boom only" sound, which was computed by convolving a measured room impulse response with the exterior excitation waveform, from each microphone measurement. In total, predicted low booms from fourteen different vehicles, each at three loudness levels, ensonified forty-two different window rattle conditions. Each measurement was repeated six times in both a lively and damped room configuration. This database of 148,176 rattle measurements enabled objective comparisons of rattle produced by different aircraft, at different booms levels and for different window rattle conditions.

2:00

4pPA2. Evaluation of the effect of aircraft size on indoor annoyance caused by sonic booms and rattle noise. Alexandra Loubeau (Structural Acoust. Branch, NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, a.loubeau@nasa.gov)

NASA plans to use a low-boom flight demonstration supersonic aircraft for community annoyance studies of sonic booms. As previously reported, laboratory studies were conducted in NASA's Interior Effects Room (IER) to help determine the relevance of using a sub-scale aircraft. Indoor annoyance was evaluated for a variety of sonic booms predicted for several different sizes of vehicles, whose levels were adjusted to the same range of loudness levels. Although no significant effect of aircraft size was found for equivalent loudness levels, a new laboratory study was conducted in the IER to extend this investigation to include the effect of secondary rattle noises. The rattle noise playback was determined from window rattle measurements performed in the IER that resulted in a database of rattle noises matched with sonic booms predicted for different aircraft, at different boom levels. The main objective was to test whether aircraft size is still not significant when realistic window rattles are included in the simulated indoor sound field. The data have also been used to evaluate a variety of noise metrics. Results using metrics computed on exterior boom levels show greater variation in subjective annoyance than in previous studies conducted without rattle.

2:20

4pPA3. Perceptual characterization of Mach cut-off sonic booms. Nicholas D. Ortega, Michelle C. Vigeant, and Victor Sparrow (Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, njo5068@psu.edu)

Mach cut-off refers to a set of flight conditions wherein the direct sound of a sonic boom does not reach the ground. This phenomenon occurs in the idealized lower atmosphere, which naturally refracts sound upwards. The current work, subdivided into two perceptual studies, investigates the perception of the evanescent sound field below the direct path of the boom. Both studies used Mach-cutoff ground signature recordings from NASA's "Farfield Investigation of No-boom Thresholds" (FaINT). In the first study, subjects provided descriptors for Mach-cutoff ground signatures in order to develop a vocabulary useful for describing these sounds. In the second study, subjects rated the stimuli on multiple scales, comprised of descriptors from the first task and annoyance, to identify which perceptual attributes factor into annoyance. These findings were used to obtain an overall perceptual characterization of the Mach-cutoff ground signatures, and these studies may inform metrics for use in predicting reactions to Mach-cutoff flyovers. [Work supported by the FAA. The opinions, findings, conclusions and recommendations expressed in this material are those of the authors and do not necessarily reflect the views of ASCENT FAA Center of Excellence sponsor organizations.]

2:40

4pPA4. How ordering effects change with cursor placements on scales when evaluating annoyance levels of transient environmental sounds. Yiyun Zhang and Patricia Davies (Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., 177 South Russell St., West Lafayette, IN 47907, zhan1728@purdue.edu)

Sound ordering effects are the impacts of the last heard sound on the rating of the current sound when evaluating a group of signals in sequence. Previously, it was shown that such effects are more significant in evaluations of transient sounds than in evaluations of steady-state sounds. The goal of this study was to investigate whether initial cursor placement on the rating scale could strengthen or attenuate ordering effects. In the previous study, the cursor was placed in the middle for the first signal, and then was left at the place from the last rating. In this follow-up test with the transient sounds, the effect of three different cursor setting strategies was examined: (1) always at the left (indicating the least annoyance level), (2) at the place of the last rating (same as the previous test), and (3) randomly placed on the rating scale. Preliminary results show that the overall trends of sound annoyance ratings with the three cursor setting strategies are similar for most subjects, and results are similar to those in the previous test. However, for some subjects, left cursor placement tends to reduce those subjects' annoyance ratings.

3:00–3:15 Break

3:15

4pPA5. A wide-area system for tracking supersonic aircraft overflights and capturing their shock waves. Gonzalo Sanchez (Sanchez Industrial Design Inc., 4319 Twin Valley Rd., Ste. 3, Middleton, WI 53562, gsanchez@sid-inc.com), Richard Horonjeff (None, Boxborough, MA), Vincent Mestre (Landrum & Brown, Irvine, CA), and Sanford Fidell (Fidell Assoc. Inc, Woodland Hills, CA)

An Internet-based, geographically-distributed prototype system for capturing sonic booms produced by NASA's low boom flight demonstrator (LBFD) aircraft has been successfully field tested. The system is designed to track supersonic LBFD flights over distances of hundreds of miles so that interviewing of overflown populations can be synchronized with the time of arrival of their sonic booms in multiple, widely separated communities. A central server archives reports of reception of the demonstrator aircraft's ADS-B signals from multiple field receivers, calculates shock wave arrival times at interviewing sites, captures and stores the acoustic waveform produced at the predicted time of arrival of the boom, coordinates interview start and stop time with sonic boom arrival times, uploads the captured waveforms on demand, and provides multiple, geographically-distributed analysts with password protected, remote access to the flight tracking and acoustic information in near-real time.

3:35

4pPA6. Multilevel analysis of recent noise social survey data including noise sensitivity. Jonathan Rathsam (NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, jonathan.rathsam@nasa.gov), Maurice E. Hayward (None, Newport News, VA), Laure-Anne Gille (None, Vaulx-en-Velin, France), Edward T. Nykaza (ERDC-CERL, Champaign, IL), and Nicole M. Wayant (ERDC-GRL, Alexandria, VA)

Noise social surveys are typified by large variation in annoyance ratings for equivalent noise exposure. Some of this variation has previously been linked to individual attitudes, including noise sensitivity. A framework provided by multilevel statistical analysis can be used to study the relationships between noise annoyance and individual attitudes. The current effort applies multilevel analysis to two recent community surveys of impulsive noise for which noise-sensitivity was assessed via a baseline survey. In the results, the sizable annoyance variation among respondents is quantified. Multilevel models with random effect terms for slope and or intercept describe more variation than multiple regression models with only fixed effects for noise exposure and noise sensitivity.

3:55

4pPA7. Determining a sufficient range of sound levels for single event analysis in quiet sonic boom community surveys. Jasme Lee (Dept. of Statistics, North Carolina State Univ., NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, jasme.lee@nasa.gov), Jonathan Rathsam (NASA Langley Res. Ctr., Hampton, VA), and Alyson G. Wilson (Dept. of Statistics, North Carolina State Univ., Raleigh, NC)

In order to enable quiet supersonic flight overland, NASA is establishing a quiet sonic boom database to present to national and international noise regulators. The database will contain dose-response models to provide evidence of a cause-and-effect relationship between boom sound levels and annoyance. In order to build the dose-response models, NASA is designing surveys for data collection. An important aspect in planning for future community surveys is the range of sound levels necessary for single event analysis. The dose-response models must be characterized at sound levels both above and below a notional threshold. The trade-offs for testing at high sound levels are the increased risk of negative community reaction and misconception of quiet sonic booms. On the other hand, testing only at low sound levels will result in unstable estimates in the dose-response models. To find a compromise between the two, analysis of a pilot community survey from 2011 explores how excluding high level booms will affect the dose-response analysis since both low and high level booms were present in the study. The data analysis will also provide insight into the appropriate range of sound levels for single event analysis in future community surveys.

Session 4pPPa

Psychological and Physiological Acoustics: Sound Localization, Spatial Release from Masking, and Binaural Hearing with Devices

Nirmal Kumar Srinivasan, Chair
Audiology, Speech-Language Pathology, and Deaf Studies, Towson University, 8000 York Road, Towson, MD 21252

Contributed Papers

1:00

4pPPa1. Improvements in transaural synthesis with the Moore-Penrose pseudoinverse matrix. Aimee Shore and William M. Hartmann (Phys. and Astronomy, Michigan State Univ., 567 Wilson Rd, East Lansing, MI 48824, shoreaim@msu.edu)

Transaural synthesis using loudspeakers is a powerful technique for conducting binaural hearing experiments while avoiding myriad problems caused by headphones. It is the most promising method to test binaural hearing for listeners using cochlear implants. Transaural synthesis is typically implemented using crosstalk cancellation, a technique whereby the left-channel signal and right-channel signal are adjusted so that reproduction by two synthesis loudspeakers results in near perfect transmission of the target left and right signals to the left and right ears (or processor microphones). Crosstalk cancellation has enabled precise experimental control over the stimulus delivered to each ear. Unfortunately, the 2×2 matrix inversion required for crosstalk cancellation can occasionally introduce spuriously large amplitudes for specific frequencies into the adjusted signals. These large amplitudes lead to perceptually salient tones. We demonstrate through simulation and experiment in a real room that adding a third loudspeaker and solving the resulting 2×3 problem using the Moore-Penrose pseudoinverse matrix yields dramatically fewer large amplitudes in the resulting adjusted waveforms. We have also conducted experiments to investigate the 2×3 system's robustness to inadvertent listener motions.

1:15

4pPPa2. Threshold interaural time differences under optimal conditions. Mathias Dietz and Sinthiya Thavam (National Ctr. for Audiol., Western Univ., 1201 Western Rd., London, ON N6G 1H1, Canada, mdietz@uwo.ca)

Klumpp and Eady (1956, J Acoust Soc Am 28, p.859-860) reported preliminary data on human sensitivity to interaural time differences (ITD) with various stimuli. At 10 μs ITD the best discrimination of 79% correct was reported for band-pass filtered (150-1700 Hz) noise. Despite the preliminary nature, and presentation methods different from todays, the above is still the best available reference for optimal ITD discrimination. The goal of the current study is to systematically determine the stimulus and the experimental paradigm that results in the smallest threshold ITD and to provide an accurate reference value. We varied seven stimulus and procedure parameters: stimulus waveform, stimulation level, stimulus duration, adaptive versus constant stimulus procedure, alternative-forced-choice (AFC) procedure, inter-stimulus pause duration, and complete waveform versus ongoing ITD. The condition yielding the lowest threshold ITD was Gaussian noise band-pass filtered from 20 to 1400 Hz, presented at 70 dB SPL, with a short inter-stimulus pause of 50 ms, and an interval duration of 0.5 s. Averaged across 8 trained subjects, the threshold ITD for this condition at the 79% correct level was 7 μs. The influence of each parameter will be discussed together with the obstacles of accurately determining this value.

1:30

4pPPa3. Importance of low frequency hearing for spatial release from masking. Nirmal Kumar Srinivasan, Maxwell Schmidt, Alexis Staudenmeier, and Kelli Clark (Audiol., Speech-Lang. Pathol., and Deaf Studies, Towson Univ., 8000 York Rd., Towson, MD 21252, nsrinivasan@towson.edu)

Listeners tend to achieve better speech understanding when the masking talker(s) are spatially separated from the target talker; an effect known as spatial release from masking. Here, we present data on spatial release from masking experiment where the target and maskers are high pass filtered with a three frequency audiometric pure-tone threshold equal to 20 dB HL. Young normal hearing listeners were presented with CRM sentences in anechoic and reverberant listening environments. The target speech was presented simultaneously with two maskers that were either colocated (0° azimuth) or symmetrically separated from the target in azimuth (±15°). Reverberant listening environments were simulated using techniques described in Zahorik (2009) and reverberation times (T60) of 1s and 2s were used. Preliminary results indicate that the amount of release from masking reduced for filtered speech and this reduction was greater for reverberant listening environments as compared to anechoic environment.

1:45

4pPPa4. Is there spatial release from listening effort in noise and reverberation? Jan Rennies and Gerald Kidd (Dept. of Speech, Lang. and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, jrennies@bu.edu)

Spatial unmasking of speech in the presence of interfering sounds has been investigated mostly by assessing speech recognition performance (e.g, SRTs or percent correct). Typically, these experiments measure recognition under very difficult listening conditions in which optimal performance (i.e., obtained in quiet) is reduced. For normal-hearing listeners this usually means presenting the signal at low/negative SNRs to avoid ceiling effects. However, such conditions may not be representative of everyday listening and hence the generalization of the results to real-world environments is limited. However, several recent studies have concluded that *listening effort* may be more meaningful than intelligibility because it can vary across a wider range of SNRs including positive SNRs where intelligibility is at ceiling. Despite this advantage, relatively few studies have examined listening effort in realistic listening conditions. In particular, binaural effects in listening effort have not been systematically investigated. The goal of this study is to measure listening effort in normal-hearing listeners in conditions with systematically varying binaural parameters, involving one or more interferers and different degrees of reverberation. Listening effort is assessed using categorical listening effort scaling, and experimental data are compared to predictions of a binaural listening effort model derived from a binaural speech intelligibility model.

2:00

4pPPa5. Binaural perceptual weighting of reverberation level in normal hearing listeners. Gregory M. Ellis (Dept. of Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY 40292, g.ellis@louisville.edu) and Pavel Zahorik (Otolaryngol. and Communicative Disord., Univ. of Louisville and Heuser Hearing Inst., Louisville, KY)

In general, perceived reverberation strength is related to physical reverberation level present at the two ears; however, this relation depends on listening condition. Pervious work using virtual auditory space techniques has demonstrated that when physical reverberation is reduced in one ear while leaving the other ear unchanged, listeners do not report a change in reverberation strength. Reducing physical reverberation equally in both ears under binaural listening or in the signal ear under monaural listening elicits a decrease in perceived reverberation strength. To better understand the relation between physical and perceived reverberation in different listening conditions, a perceptual weighting experiment was performed. Listeners reported perceived reverberation strength in a test stimulus using a magnitude estimate paradigm relative to a standard while listening over headphones. Test stimuli were generated by jittering reverberation level independently in the two channels of a binaural room impulse response (BRIR) before convolution with a speech token. BRIRs with reverberation levels of -12, -6, and 0 dB at each ear relative to the standard were tested. Results of a perceptual weighting analysis suggest that listeners use different weighting strategies in different listening conditions, and are generally consistent with a "better ear" hypothesis for reverberation listening.

2:15–2:30 Break

2:30

4pPPa6. Using binaural beat sensitivity to explore mechanisms of bimodal temporal envelope beat sensitivity. Coral Dirks, Peggy Nelson, and Andrew J. Oxenham (Univ. of Minnesota, 164 Pillsbury Dr SE, Minneapolis, MN 55455, hans3675@umn.edu)

Current cochlear implant (CI) fitting strategies aim to maximize speech perception through the CI by allocating all spectral information across the electrode array without regard to tonotopic placement of each electrode along the basilar membrane. For patients with considerable residual hearing in the non-implanted ear, this approach may not be optimal for binaural hearing. This study aims to explore fitting procedures in which CI maps better complement information from the acoustic ear by reducing the frequency mismatch between them. We investigate the mechanisms of binaural temporal-envelope beat sensitivity in normal-hearing listeners using bandpass filtered pulse trains with parameters including stimulus level, filter bandwidth, filter slope, and spectral overlap using bandpass filtered pulse trains. We find the minimum baseline interaural timing difference and spectral mismatch that normal-hearing listeners can tolerate while maintaining their ability to detect interaural timing differences. Initial results consistently demonstrate maximum sensitivity to binaural beats when place of stimulation is matched across ears. The outcomes of this study will provide new information on binaural interactions in normal-hearing listeners and guide methodology for incoming single-sided-deafness patients as we adjust their CI maps in an effort to reduce the frequency-mismatch. [Work supported by NIH grant F32DC016815-01.]

2:45

4pPPa7. Impact of the interaural stimulation timing mismatch on localization performance in bimodal HA/CI users. Stefan Zirn, Julian M. Angermeier (Elec. Eng. and Information Eng., Univ. of Appl. Sci. Offenburg, Badstraße 24, Offenburg 77652, Germany, stefan.zirn@hs-offenburg.de), and Thomas Wesarg (ENT, Medical Ctr. of the Univ., Freiburg, Germany)

The normal-hearing human auditory system is able to perceive interaural time differences (ITD) as small as 10 μs. The largest ITD occurring physiologically is about 700 μs. In bimodal users differences in processing latencies of digital hearing aids (HA) and cochlear implants (CI) up to 9 ms superpose these tiny ITD resulting in an interaural stimulation timing mismatch. Our hypothesis in the present study is that this interaural stimulation timing mismatch impairs sound localization in bimodal HA/CI users. To investigate, we conducted localization tests in bimodal users with and without differences in processing latencies. To compensate for individual processing latency differences we designed a wearable programmable delay line based on a circular buffer implemented on a microcontroller in order to delay the faster device (the CI system). In each subject, an initial localization test with the delay line deactivated was conducted. Afterwards the delay line was activated followed by a 1 hour familiarization period. Finally, the localization test was repeated. Results showed an improvement in sound localization in the horizontal plane of 10 % averaged across 8 bimodal users after compensation. The effect was significant ($p<.05$) using a Wilcoxon signed rank test.

3:00

4pPPa8. Binaural speech unmasking and interference in adult bilateral cochlear-implant users. Matthew Goupell, Olga Stakhovskaya (Hearing and Speech Sci., Univ. of Maryland, College Park, 0119E Lefrak Hall, College Park, MD 20742, goupell@umd.edu), and Joshua G. Bernstein (Walter Reed National Military Medical Ctr., Bethesda, MD)

Bilateral cochlear implants (BICIs) provide improved speech perception in noise, primarily derived from the head-shadow benefit. BICI listeners can also demonstrate binaural speech unmasking or squelch, particularly if the interferer is a single other speaker. However, under the same conditions, some BICI listeners can experience interference. This study tested 21 adult BICI listeners on speech understanding using a range of masker conditions (1, 2, 3, or 4 same-sex talkers, speech-modulated noise, or speech-shaped stationary noise). For listeners that showed binaural unmasking, the amount of unmasking was highest for the one-talker condition and least for the stationary noise, decreasing as the number of interfering talkers increased and the envelope modulation depth decreased. For listeners that showed interference, there were mainly two types of interference patterns: (1) where the amount of interference varied idiosyncratically across masker type, and (2) any stimulation in one ear essentially produced chance performance in the other ear for all masker conditions. The latter interference pattern occurred for the listeners with the largest asymmetries in speech understanding. This suggests, in these extreme cases, a possible central cortical processing bias for one ear and a more equal speech understanding between the ears may diminish interference.

3:15

4pPPa9. Evaluating sound localization cues for premium hearing aids across multiple manufacturers. Anna C. Diedesch (Commun. Sci. & Disord., Western Washington Univ., 516 High St., MS 9171, Bellingham, WA 98225, anna.diedesch@wwu.edu), G. Christopher Stecker (Vanderbilt Univ., Nashville, TN), and Frederick J. Gallun (VA RR&D National Ctr. for Rehabilitative Auditory Res., Portland VA Health Care System, Portland, OR)

Modern hearing aids employ complex signal processing to improve signal-to-noise ratios for people with impaired hearing. Some of this processing, however, can reduce interaural level differences (ILD), which could reduce sound localization abilities and thus interfere with communication in complex acoustic scenes. It has been speculated that interaural time differences (ITD) are also distorted by amplification, perhaps due to even very brief signal processing delays or multi-path acoustics. Previously we found little ITD distortion for a single set of hearing aids, except when "strong" directional microphones were enabled. To extend and replicate these results, premium hearing aids from four major manufacturers were compared. Aids were set to low-gain amplification and a range of different signal processing algorithms were systematically activated in a manner consistent with clinical practice. Binaural acoustical recordings were measured using an acoustic manikin on which the hearing aids were attached as would be the case for a typical patient. This presentation will describe recordings collected in anechoic and simulated rooms for which frequency-specific ITD and ILD have been extracted and compared to unaided recordings. [Work supported by the F.V. Hunt Postdoctoral Research Fellowship, NIH R01-DC011548 (GCS), R01-DC011828 (FJG), and the VA RR&D NCRAR.]

Session 4pPPb

Psychological and Physiological Acoustics: Clinical Topics and Devices (Poster Session)

Tanvi D. Thakkar, Chair
Communication Sciences and Disorders, University of Wisconsin-Madison, 934B Eagle Heights Dr., Madison, WI 53705

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

Contributed Papers

4pPPb1. The optimal noise-rejection threshold for normal and impaired hearing. Jordan Vasko, Eric Healy (Speech & Hearing Sci., The Ohio State Univ., Pressey Hall Rm 110, 1070 Carmack Rd, Columbus, OH 43210, healy.66@osu.edu), and DeLiang Wang (Comput. Sci. and Eng., The Ohio State Univ., Columbus, OH)

Binary masking represents a powerful tool for increasing speech intelligibility in noise. An essential aspect involves the local criterion (LC), which defines the signal-to-noise ratio below which time-frequency units are discarded. But binary masking is a victim of its own success in one regard—it produces ceiling sentence intelligibility across a broad range of LC values, making the exact optimal LC value difficult to determine. Further, the optimal value for hearing-impaired (HI) listeners is largely unknown. In the current study, the optimal LC was determined in normal-hearing (NH) and HI listeners using speech materials less likely to produce ceiling effects. The CID W22 words were mixed with noise consisting of recordings from a busy hospital cafeteria, then subjected to ideal binary masking. LC values ranged from -20 to + 5 dB relative to the overall SNR of -8 dB. NH subjects were tested at 65 dBA and HI subjects were tested at 65 dBA plus NAL-RP hearing-aid gains. Preliminary results suggest that the optimal LC is similar for NH and HI listeners. Additional conditions involving different speech materials and noise types suggest that the optimal LC can vary as a function of speech and/or noise type. [Work supported by NIH.]

4pPPb2. Effects of self-generated noise on quiet threshold by transducer type in school-age children and adults. Heather Porter, Lori Leibold (Boys Town National Res. Hospital, 555 N. 30th St., Omaha, NE 68131, heather.porter@boystown.org), and Emily Buss (Univ. of North Carolina at Chapel Hill, Chapel HIll, NC)

This study tested the hypothesis that higher levels of self-generated noise in children results in larger child-adult differences for the detection of low-frequency sounds when testing is conducted using transducers associated with a pronounced occlusion effect. Detection thresholds were measured at 125, 250, 500, and 1000 Hz using standard clinical procedures with supra-aural headphones, insert earphones, or a loudspeaker. Probe microphone recordings were made during testing with each transducer. Listeners were 4.5- to 11-year-olds, and adults. Preliminary results are consistent with the a priori hypothesis. For all listeners, thresholds at 125 and 250 Hz were highest with supra-aural headphones and lowest with free-field presentation. This transducer effect was most pronounced for younger children. Child-adult differences were smaller at 500 and 1000 Hz, an observation consistent with reduced effects of self-generated noise with increasing frequency. Trial-by-trial analysis of probe microphone recordings will be presented.

4pPPb3. Integrating remote microphone signals with hearing-aid processing. James Kates and Kathryn Arehart (Speech Lang. Hearing Sci., Univ. of Colorado, 409 UCB, Boulder, CO 80309, James.Kates@colorado.edu)

Remote microphones (RM) are used with hearing aids to provide speech signals free from noise and room reverberation. However, because the RM signal bypasses the acoustics of the head and pinna, it can provide an unnatural signal that is lateralized at the center of the head rather than localized at the talker. A structural binaural model is combined with synthesized early room reflections to provide the missing externalization cues. The effectiveness of this simulation has been demonstrated over headphones using linear amplification to compensate for the hearing loss. In a hearing aid, however, the RM processed signal is combined with the acoustic inputs at the hearing-aid microphones, and the processed hearing-aid output is combined with the reverberated speech that is transmitted through the vent or open fitting. The hearing-aid processing implemented in the experiments described in this presentation includes both linear amplification and wide dynamic-range compression. The externalization, spatial soundfield characteristics, and sound quality provided by the entire signal-processing system are evaluated for normal-hearing and hearing-impaired listeners.

4pPPb4. Assessing perceived listening difficulty using behavioral gaze patterns for audiovisual speech. Sterling W. Sheffield and Joshua G. Bernstein (Walter Reed National Military Medical Ctr., 4954 North Palmer Rd, Bethesda, MD 20889, sterling.sheffield.ctr@mail.mil)

Listeners alternate their gaze between the talker's eyes, nose, and mouth during conversation, spending more time gazing directly at the mouth in more difficult listening conditions. Perceived listening difficulty and speech perception performance vary independently, demonstrated with indirect measures of perceived listening difficulty/effort such as pupil dilation and dual task paradigms. This study took a more direct approach, evaluating whether perceived listening difficulty could be assessed by observing a listener's natural gaze pattern in a challenging speech-perception task. The proportion of time spent gazing at the talker's mouth was measured for auditory-visual speech perception in conditions that produced equal percentage-correct scores, but where one condition was designed to produce greater perceived difficulty. Experiment 1 compared gaze proportion for a speech-shaped noise versus two same-gender interfering talkers. Experiment 2 examined the effect of sentence context using low and high-probability sentences. The results are discussed in terms of the potential application of the eye-gaze metric to evaluate rehabilitative interventions in cases where subjectively reported benefits do not manifest in improved speech-understanding scores. [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.]

4pPPb5. Alternative methods for faster and more accurate estimation of hearing threshold. Amitava Biswas and Jennifer Goshorn (Speech and Hearing Sci., Univ. of Southern MS, 118 College Dr. #5092, USM-CHS-SHS, Hattiesburg, MS 39406-0001, Amitava.Biswas@usm.edu)

Determination of audiologic threshold is often necessary for variety of diagnoses and treatments. The usual method involves "10 dB down for response and 5 dB up for miss". This common procedure lacks sufficient accuracy to identify smaller changes due to temporary threshold shift and for early diagnosis of degradation of hearing threshold due to ototoxicity or chemotherapy etc. This presentation will explore several alternative methods for faster and more accurate estimation of the threshold.

4pPPb6. Exploring some questions on occlusion or the Bing effect during speech production with a microphone coupled to the speaker's external ear canal. Amitava Biswas (Speech and Hearing Sci., Univ. of Southern MS, 118 College Dr. #5092, USM-CHS-SHS, Hattiesburg, MS 39406-0001, Amitava.Biswas@usm.edu)

Sometimes some speakers prefer to use their palm or other objects to cover or occlude at least one ear during public address. This practice may be helpful to enhance their self monitoring of the sound production using the occlusion effect. The basic occlusion effect in the human auditory system has been explored and reported in the literature by several individuals. According to those reports, the speaker can hear his or her own voice significantly louder when the ear canal is blocked at the outer end. Many clinicians routinely utilize detectability of vibrations from a tuning fork when placed on the mastoid process and the ear canal is occluded. This is popularly known as the Bing effect. These empirical data suggest existence of an efficient acoustic connectivity from the vocal tract to the ear canal for healthy individuals. Therefore, this study will explore quantified effects of the classic Bing effect for several speakers across the audio spectrum with a microphone coupled to the speaker's external ear canal.

4pPPb7. Online training for perceptual rehabilitation in cochlear implant users. Damian Almaraz, Elizabeth Een, Elaine Grafelman (Psych., St. Olaf College, Northfield, MN), Justin Pacholec (Comput. Sci., St. Olaf College, Northfield, MN), Yadi Quintanilla, Paula Rodriguez, and Jeremy Loebach (Psych., St. Olaf College, 1520 St. Olaf Ave., Northfield, MN 55057, loebach@stolaf.edu)

Although cochlear implants have been demonstrated to be effective surgical treatments for deafness, new implant users must undergo an intense period of perceptual learning and adaptation to learn to hear with their prosthesis. Few adult cochlear implant users receive any formal training following implantation, and as a result, the perceptual skills that they develop are highly variable. This study assessed the efficacy of the high variability online training program for adult cochlear implant users dubbed the HiVOlT-CI in 31 normal hearing individuals listening to cochlear implant simulations and three adult cochlear implant users. The goal of our interactive, adaptive, high-variability training program is to provide empirically-based, adaptive and interactive perceptual training to new cochlear implant users to help them adjust to their prosthesis. We also aim to develop a common set of robust and adaptive cognitive auditory skills in cochlear implant users that will help them to hear in real world situations. We also hope to establish baseline levels of performance for experienced cochlear implant users to evaluate successful implant use by new users.

4pPPb8. The development of sentence recognition in noise by cochlear implant users. Chris J. James, Chadlia Karoui, Mathieu Marx, Marie-Laurence Laborde, Marjorie Tartayre, Olivier Deguine, and Bernard Fraysse (ORL, Hôpital Purpan, Pl. du Dr Baylac, Toulouse 31300, France, chris.j.james@wanadoo.fr)

We previously established that under certain physiological conditions the sentence recognition scores in noise of cochlear implant (CI) users one month after activation are similar to those for normal listeners listening to 10–12 channel acoustic (vocoder) simulations (James *et al.*, CIAP, Lake Tahoe, July 2017). We further analysed sentence recognition scores in noise over time in order to determine whether short term, 1-month scores predict long-term outcomes for sentence recognition in noise. Long-term sentence-in-noise data for fixed-SNR-level testing (10, 5, 2.5, 0 dB) were reduced to SNR50 values using a validated analytic sigmoid fit. Data were analysed for 106 adult CI users. 12-month SNR50s were highly correlated with 1-month sentence recognition scores in quiet and 10 dB SNR ($r^2 = 0.29$ and $r^2 = 0.22$, both $p<0.001$): CI users with initially poorer scores could catch up to some extent with early, better performers. However, early high performance (>80% correct score in quiet) was generally associated with 12-months SNR50s $\leq$2 dB. Early sentence recognition scores for CI users may predict their long-term recognition of speech in noise. The number of effective channels available to CI users may evolve to greater than 10-12 in the long term. Disclosure: C.J. is also an employee of Cochlear France. C.K. received partial doctoral funding from Cochlear France as part of the program "Conventions industrielles de formation par la recherche" (CIFRE).

4pPPb9. Binaural intelligibility level difference with a mixed-rate strategy simulation. Thibaud Leclère, Alan Kan, and Ruth Litovsky (Waisman Ctr., Univ. of Wisconsin-Madison, 1500 Highland Ave., Waisman Ctr., Madison, WI 53705, leclere2@wisc.edu)

In noisy situations, bilateral cochlear implant (BiCI) listeners demonstrate poorer spatial release from masking compared to normal hearing (NH) listeners. One reason for this difference could be the limited access for BiCI listeners to fine structure interaural time differences (ITDs) which are crucial for binaural unmasking. Clinical processors convey speech envelopes using pulsatile high-rate (~1000 Hz) carriers, and discard the original temporal fine structure. At high rates, BiCI listeners are not sensitive to ITDs, whereas low rates (< 300 Hz) result in better ITD sensitivity. To preserve both ITD sensitivity and well-sampled speech envelopes, high and low rates are needed. The present study examines whether a mixed rate strategy will enable binaural unmasking in NH listeners using a 16-channel mixed-rate vocoder. Binaural intelligibility level differences were calculated from speech reception thresholds measurements in five mixed-rate configurations. Targets had a 0-μs ITD and masking speech-shaped noises were co-located or separated by applying a 400-μs ITD to the low-rate channels. Results from this experiment will help determine the potential benefits of a mixed-rate strategy to restore spatial unmasking abilities of CI listeners in noisy situations.

4pPPb10. The role of onset cues for lateralization of an auditory object in bilateral cochlear implant listeners. Tanvi D. Thakkar, Alan Kan, and Ruth Litovsky (Commun. Sci. and Disord., Univ. of Wisconsin-Madison, 934B Eagle Heights Dr., Madison, WI 53705, tthakkar@wisc.edu)

In bilateral cochlear implant (BiCI) listeners, a common strategy for improving sensitivity to interaural timing differences (ITDs) is to introduce low rate stimulation in some channels. Previous work showed that BiCI listeners can lateralize a single sound with a constant ITD at every pulse of a low rate stimulus. However, in reverberant environments, ITDs for a target sound can change over a short period of time. Previously we have shown that BiCI listeners had difficulty using the ITD in the first pulse of a low-rate pulse train to lateralize a stimulus when the subsequent pulses had ITDs that varied over time. Hence, we hypothesized that a more salient onset is needed in order to restore ITD sensitivity in BiCI listeners when using low rate stimulation. In this study, we measured the number of pulses with a consistent ITD that is needed at the onset in order for BiCI listeners to reliably perceive one sound in the correct location. Results suggest that BiCI listeners need at least five pulses with consistent ITDs in order to report a single auditory object that is correctly lateralized. These results have important implications for low rate strategies aimed at restoring ITD sensitivity to BiCI listeners.

4pPPb11. The effect of movement, compression, and cochlear-implant processing on the output of cochlear implants. Alan Archer-Boyd and Robert P. Carlyon (Univ. of Cambridge, 15 Chaucer Rd., Cambridge CB2 7EF, United Kingdom, alan.archer-boyd@mrc-cbu.cam.ac.uk)

Dynamic-range compression is used in cochlear implants (CIs) to compensate for the greatly reduced dynamic range of CI listeners. It has been found that unlinked bilateral compression in hearing aids (HAs) can affect the spatial location of sounds, reducing interaural level differences (ILDs)

and causing sound sources to appear closer to 0° azimuth and/or more diffuse (Wiggins and Seeber, 2011). The compressors in CI processors are often much stronger than hearing aids (e.g., CI compression ratios of 12:1 vs. ≤3:1 in HAs). We investigated the effect of compression on monaural and binaural level cues during simulated rotational movements in the horizontal plane. Moving signals were created by combining static behind-the-ear hearing aid microphone impulse responses at successive angles. For sound levels above the compressor threshold, fast movements over wide angles (120°/s) altered the size and extent of monaural level changes and ILDs considerably, relative to those from those that would occur naturally and without compression. The relatively slow attack and release times of the compressor also had the effect of producing a "lag" in the level changes after the movement had stopped. Microphone directionality was also found to have an effect on the output of the compressor during movement.

4pPPb12. The segregation of sequentially interleaved sentences by normal-hearing and cochlear-implant listeners using monaural level cues and the effect of dynamic-range compression. Alan Archer-Boyd and Robert P. Carlyon (Univ. of Cambridge, 15 Chaucer Rd., Cambridge CB2 7EF, United Kingdom, alan.archer-boyd@mrc-cbu.cam.ac.uk)

Normal-hearing (NH) listeners can use level differences to segregate pairs of vocoded sentences that are presented to interleaved frequency channels; performance is better at both positive and negative target-to-masker ratios (TMRs) compared to a TMR of 0 dB (Ihlefeld and Shinn-Cunningham, 2008). The present study investigated whether unilateral cochlear-implant (CI) users could also use level differences alone to segregate two sentences. Due to the greatly reduced frequency resolution and increased TMR thresholds of CI users, the sentences were interleaved in time and presented sequentially. The Boston University (Kidd *et al*., 2008) corpus was used to construct interleaved, five-word sentences, consisting of name, verb, number, adjective, and object. The presentation order of the words alternated randomly for each word type. Each sentence began with the same two words and listeners were required to select the last three words in the target sentence. The level differences ranged from -12 to 12 dB. A monaural vocoder study showed that the task was possible for both with younger and older NH listeners, with performance increasing as level difference increased. The results of the CI user study will be presented, and the effects of dynamic-range compression on their performance discussed.

4pPPb13. Effects of broad binaural fusion and hearing loss ondichotic concurrent vowel identification. Lina Reiss (Otolaryngol., Oregon Health & Sci. Univ., 3181 SW Sam Jackson Park Rd., Mailcode NRC04, Portland, OR 97239, reiss@ohsu.edu), Sara Simmons, Larissa Anderson (School of Audiol., Pacific Univ., Portland, OR), and Michelle R. Molis (National Ctr. for Rehabilitative Auditory Res., Portland Veterans Administration, Portland, OR)

Many hearing-impaired (HI) individuals have abnormally broad binaural pitch fusion, such that tones differing in pitch by up to 3-4 octaves are fused between ears (Reiss *et al*., 2017). Broad binaural fusion leads to averaging of the monaural pitches (Oh and Reiss, 2017), and may similarly lead to fusion and averaging of speech streams across ears. In this study, we examined the relationship between binaural fusion range and dichotic vowel identification in normal-hearing (NH) and HI listeners. Synthetic vowels /i/,/u/,/a/, and /ae/ were generated with three fundamental frequencies (F_0) of 106.9, 151.2, and 201.8 Hz, and presented dichotically through headphones. For HI listeners, stimuli were shaped according to NAL-NL2 prescriptive targets. Listeners identified 1 vowel or 2 different vowels on each trial. Preliminary results indicate that NH listeners improved their percentage of both vowels identified as the ΔF_0 increased, but HI listeners did not. HI listeners were more likely to fuse two vowels together even with large ΔF_0. NH listeners with broad fusion also had poorer overall performance than those with sharp fusion. The findings suggest that broad binaural fusion can impede separation of voices and speech streams in both NH and HI listeners. [Work supported by NIH-NIDCD grant R01 DC013307.]

4pPPb14. Mechanisms underlying speech masking release in hybrid cochlear implant users. Viral Tejani (Dept. of Otolaryngology-Head & Neck Surgery, Univ. of Iowa Hospitals and Clinics, 200 Hawkins Dr., 21003 PFP, Iowa City, IA 52242, viral-tejani@uiowa.edu) and Carolyn Brown (Commun. Sci. & Disord., Univ. of Iowa, Iowa City, IA)

Hybrid cochlear implant (CI) users utilize low-frequency acoustic and high-frequency electric stimulation in the implanted ear, outperform traditional CI users in speech perception in noise (Gantz *et al*., 2017), and show "masking release," an improvement in speech recognition in fluctuating noise relative to steady noise (Turner *et al*., 2004). Improved performance in noise has been attributed (but not proven) to the spectral resolution and temporal fine structure (TFS) provided by acoustic hearing. Spondee recognition in two-talker and ten-talker babble was assessed in Hybrid CI users, and masking release was calculated as the percent difference in performance in two-talker (easy condition) and ten-talker (hard condition) babble. Spectral ripple density discrimination thresholds and fundamental frequency discrimination limens was also assessed, as these psychophysical measures reflect underlying spectral resolution and TFS and correlate with speech perception in noise in traditional CI users (Won *et al*., 2007; Goldsworthy *et al*., 2013; Goldsworthy, 2015; Jeon *et al*., 2015). Preliminary data suggest masking release is possible in Hybrid CI users, and that psychophysical measures and masking release may be correlated, suggesting a role of spectral resolution and TFS in masking release. The extent of residual hearing also appears to correlate with psychophysical and masking release performance.

4pPPb15. Modulation interference in cochlear implants. Monita Chatterjee and Aditya M. Kulkarni (Boys Town National Res. Hospital, 555 N 30th St, Omaha, NE 68131, monita.chatterjee@boystown.org)

Modulation interference, in which the temporal envelope of a signal is rendered less salient due to the presence of a competing temporal envelope of a masker, has been modeled as the result of modulation-tuned filters operating in the midbrain. In listeners with cochlear implants (CIs) who rely more heavily on the speech temporal envelope, such interference may be particularly problematic for speech perception in competing, fluctuating backgrounds. Here, we present evidence for modulation interference in CI users from our laboratory, using both non-speech and speech targets. Maskers were presented either in the same ear or in the contralateral ear relative to the target, and either concurrently with the target or preceding the target (forward masking). Tasks include the detection of the temporal envelope on the target as well as the discrimination of temporal envelopes between targets. Results suggest that modulation interference plays an important role in CI users, reducing the salience of the speech temporal envelope under a variety of masking conditions.

4pPPb16. Improving temporal modulation sensitivity in cochlear implant users. Shaikat Hossain (Otolaryngol., Univ. of Southern California, 7220 Mccallum Blvd #307, #307, Dallas, TX 75252, shossa3@gmail.com), Susan R. Bissmeyer (Biomedical Eng., Univ. of Southern California, Newhall, CA), and Raymond L. Goldsworthy (Otolaryngol., Univ. of Southern California, Los Angeles, CA)

The present study investigated the limitations of temporal modulation sensitivity in cochlear implant (CI) users. Adult users of Cochlear Nucleus devices were tested by presenting sounds through their clinical processors using an auxiliary cable and using direct electrical stimulation provided through a custom research processor. Threshold and comfort current levels were determined by adjusting the phase duration of 10 Hz amplitude modulated pulse trains of 400 ms duration at a given stimulation rate. After mapping the participants, modulation detection thresholds were measured by means of a 3 alternative forced choice procedure which adaptively adjusted the modulation depth to determine the threshold in dB relative to full modulation. Participants completed the task using a graphical user interface to indicate which interval was different. Modulation frequencies from 10 to 800 Hz were tested at various stimulation rates on basal and apical electrodes in both psuedomonopolar and bipolar modes of stimulation. Preliminary findings indicate that modulation sensitivity drops off with increases in modulation frequency. Performance was significantly better with direct stimulation as compared to through the clinical processor and better at

higher modulation frequencies than previous findings from CI users. Outcomes will have implications for the development of next generation CI devices.

4pPPb17. Comparison of different implementations of the ACE cochlear implant signal processing strategy with an objective metric. Bernardo Murta (Federal Univ. of Santa Catarina, Rua da Passagem, 111, Belo Horizonte, Minas Gerais 30220-390, Brazil, be.murta@gmail.com), Rafael Chiea, Gustavo Mourão (Federal Univ. of Santa Catarina, Florianopolis, Brazil), Stephan Paul (Federal Univ. of Santa Catarina, Joinville, Brazil), and Julio A. Cordioli (Federal Univ. of Santa Catarina, Florianópolis, SC, Brazil)

Research that seeks to compare outcomes of Cochlear Implant (CI) processing strategies must ensure that different implementations of a given strategy are safe, and means for their comparison are required. This work aims to analyze the sensitivity of an objective metric to the effects introduced by different sets of filter bank models, compressing functions, envelope detectors and other aspects in implementations of the same CI signal processing strategy. To this end a subset of signals containing pure tones, complex harmonics, exponential sweeps and speech signals is defined and processed by means of the same CI strategy (ACE) in two different implementations: a standard version from the Cochlear Nucleus Toolbox and another version implemented by the authors using published data (FSC). As expected, variations in the output are due to different aspects in the signal processing chain of each implementation. While the proposed metric is sensitive to this differences, its magnitude also depends heavily on the test signals used and specificity is very low. Therefore a more accurate, specific and sensitive protocol was developed using UTD's CCi-Mobile Research Interface to analyze the root of the differences. This includes a metric and performance criteria that are specific for certain signal processing parameters.

4pPPb18. Mitigating the effects of reverberation and noise in cochlear implants. Kevin M. Chu (Biomedical Eng., Duke Univ., 101 Sci. Dr., Durham, NC 27708, kmc100@duke.edu), Chandra S. Throckmorton, Leslie M. Collins, and Boyla O. Mainsah (Elec. and Comput. Eng., Duke Univ., Durham, NC)

In listening environments with room reverberation and background noise, cochlear implant (CI) users experience substantial difficulties in understanding speech. Because everyday environments have different combinations of reverberation and noise, there is a need to develop algorithms that can mitigate both effects to improve speech intelligibility. Desmond *et al.* (2014) developed a machine learning approach to mitigate the adverse effects of late reverberant reflections of speech signals by using a classifier to detect and remove affected segments in CI pulse trains. In this study, we investigate the robustness of the reverberation mitigation algorithm in environments with both reverberation and noise. We conducted sentence recognition tests in normal hearing listeners using vocoded speech with unmitigated and mitigated reverberant-only or noisy reverberant speech signals, across different reverberation times and noise types. Improvements in speech intelligibility were observed in mitigated reverberant-only conditions. However, mixed results were obtained in the mitigated noisy reverberant conditions as a reduction in speech intelligibility was observed for noise types whose spectra were similar to that of anechoic speech. Based on these results, the focus of future work will be to develop a context-dependent approach that activates different mitigation strategies for different acoustic environments. [Research supported by NIH grant R01DC014290-03.]

4pPPb19. Band importance functions of listeners with sensorineural hearing impairment. Sarah E. Yoho (Communicative Disord. and Deaf Education, Utah State Univ., 1000 Old Main Hill, Logan, UT 84321-6746, sarah.leopold@usu.edu) and Adam K. Bosen (Boys Town National Res. Hospital, Omaha, NE)

The Speech Intelligibility Index (SII) includes a series of band importance functions for calculating the estimated intelligibility of speech under various conditions. Band importance functions describe the relative contribution of discrete spectral regions of speech to overall speech intelligibility. They are generally measured by filtering the spectrum into a series of frequency bands and determining intelligibility when each band is present, relative to intelligibility when other bands are present. The functions found in the SII are commonly used to provide estimations of intelligibility for listeners with hearing loss, even though they were developed with the use of normal-hearing listeners. In the current study, band importance functions were derived for individual listeners with sensorineural hearing impairment and a group of normal-hearing control participants. Functions were measured by filtering sentences to contain only random subsets of bands on each trial, and regressing speech recognition against the presence or absence of each band across trials. Preliminary results indicate that the shape of hearing loss plays an important role in the shape of band importance functions, with some listeners who have high-frequency sloping losses displaying decreased and even negative importance of high-frequency speech bands.

4pPPb20. Utilizing cross-channel features for reverberation mitigation in cochlear implants. Lidea K. Shahidi, Chandra S. Throckmorton, Leslie M. Collins, and Boyla O. Mainsah (Elec. and Comput. Eng., Duke Univ., 100 Sci. Dr., Durham, NC 27710, lidea.shahidi@duke.edu)

Cochlear implant (CI) users experience difficulties in understanding speech in reverberant environments as both active speech and non-informative speech segments of the reverberant signal are incorporated into the CI stimulus pattern. The non-informative speech segments are due to late reverberant signal reflections and are referred to here *as overlap-masking* segments. Desmond *et al.* (JASA, 2014) showed that removal of all overlap-masking segments from the CI stimulus pattern significantly improved the intelligibility of reverberant speech in CI users. Desmond (2014, Ph.D. dissertation) proposed a reverberation mitigation strategy based on training a machine learning classifier to identify and delete overlap-masking segments in CI pulse trains. The mitigation strategy employed *within-channel features* to mitigate overlap-masking effects in a given electrode channel. However, since adjacent electrode channels tend to enter the overlap-masking state concurrently, additional information about the state of other channels might provide useful information for classification. In this work, we explore the use of *cross-channel features* to improve classification performance. We find significant improvements in classification performance in low-frequency electrode channels after the addition of cross-channel features. Preliminary results from normal-hearing listeners performing speech recognition tests using vocoded speech under simulated reverberation will be reported. [Research supported by NIH grant R01DC014290-03.]

4pPPb21. Estimating the relative interaural place mismatch for bilateral cochlear-implant listeners. Olga A. Stakhovskaya (Hearing and Speech Sci., Univ. of Maryland, College Park, 4954 North Palmer Rd., Bldg. 19, R. 5607, Bethesda, MD 20889, olga.stakhovskaya.ctr@mail.mil), Joshua G. Bernstein (Audiol. and Speech Ctr., Walter Reed National Military Medical Ctr., Bethesda, MD), Kenneth K. Jensen (Audiol. and Speech Ctr., Walter Reed National Military Medical Ctr., Eden Prairie, MN), and Matthew Goupell (Hearing and Speech Sci., Univ. of Maryland, College Park, College Park, MD)

Frequency-matched bilateral input is important for the optimum encoding of binaural cues to facilitate sound localization, auditory scene analysis, and speech understanding in noise. Current bilateral cochlear-implant (CI) programming procedures do not account for potential interaural place-of-stimulation mismatch. This study investigated whether a perceptual test of interaural-time-difference (ITD) sensitivity can effectively estimate relative interaural mismatch for bilateral CI listeners. Ten bilateral CI listeners were tested on a two-interval left-right ITD discrimination task. Loudness balanced 300-ms, 100 pulse-per-second constant-amplitude pulse trains were delivered to single-electrode pairs using direct stimulation. Measurements were made for five reference electrodes evenly distributed along the array in one ear, and for at least five closely spaced comparison electrodes in the other ear for each reference electrode. The pair with the smallest thresholds was assumed to be the most closely place matched. ITD sensitivity was variable across listeners, with thresholds ranging from 80-2000 μs. ITD tuning curves varied across and within listeners, ranging from widths of approximately one (sharp tuning) to eight electrodes (broad tuning). While cases with narrower tuning provide clear suggestions for frequency programming to minimize interaural mismatch, estimates of mismatch were difficult to ascertain for cases with broad tuning.

Session 4pPPc

Psychological and Physiological Acoustics: Sensitivity to Spectral and Temporal Information

Christian Stilp, Chair
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Contributed Papers

3:45

4pPPc1. Infants' use of amplitude modulation cues in the segregation of concurrent vowels. Monika-Maria Oster and Lynne A. Werner (Speech and Hearing Sci., Univ. of Washington, 1417 North East 42nd St, Seattle, WA 98105, mmoster@uw.edu)

Infants have difficulties separating speech from competing sounds. One explanation is that infants do not use acoustic cues to sound source segregation as adults do. This study investigated 3- and 7-month-old infants' and adults' ability to use AM cues in concurrent vowel segregation. Seven American-English vowels were produced by a male and by a female talker. One male and one female vowel were randomly chosen and superimposed. A train of such vowel pairs was presented to listeners, who were trained to respond to the male target vowel /i:/ or /u:/. Target-to-nontarget ratios were chosen to equate performance across age groups. In the baseline condition, all vowels had flat envelopes. In the cue condition, male vowels were modulated with three envelopes from infant-directed vowels. The proportion of 3-month-old infants achieving an 80% correct criterion at a ratio of +15dB was higher in the cue than in the baseline condition. For 7-month olds and adults d' could be estimated based on 15 target and 15 nontarget trials in each condition. Preliminary results indicate that both 7-month-old infants and adults have a higher d' in the cue than in the baseline condition. Thus, infants appear to use AM to segregate concurrent vowels.

4:00

4pPPc2. Effects of age on the discrimination of modulation type for 2- and 10-Hz rates. Brian C. Moore, Sashi Mariathasan (Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB3 9LG, United Kingdom, bcjm@cam.ac.uk), and Aleksander Sek (Inst. of Acoust., Adam Mickiewicz Univ., Poznan, Poland)

Frequency modulation (FM) at a rate of 10 Hz may be detected via conversion of FM to amplitude modulation (AM) in the cochlea, while 2-Hz FM may be detected partly using temporal fine structure (TFS) information. Greater age may impair the processing of TFS information while sparing the processing of AM information. To test these ideas, while controlling for the effects of detection efficiency, a two-stage experiment was conducted. In stage 1, psychometric functions were measured for the detection of AM alone and FM alone imposed on a 1000-Hz carrier, using 2- and 10-Hz rates. In stage two, the task was to discriminate AM from FM at the same modulation rate when the detectability of the AM alone and FM alone was equated. For young normal-hearing subjects, discrimination was markedly better for the 2-Hz than for the 10-Hz rate, consistent with the idea that FM is coded in a different way from AM at 2 Hz. For older subjects with normal hearing at 1000 Hz, a similar pattern was obtained, but discrimination of AM from FM at 2 Hz was somewhat poorer than for the young normal-hearing subjects, consistent with an age-related reduction in ability to use TFS cues.

4:15

4pPPc3. Natural signal statistics and the timecourse of spectral context effects in consonant categorization. Christian Stilp and Ashley Assgari (Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu)

Speech perception is heavily influenced by surrounding sounds. When spectral properties differ between earlier (context) and later (target) sounds, this can produce *spectral contrast effects* (*SCEs*) that bias categorization of later sounds. For example, when context sounds have a low-F3 bias, listeners report more high-F3 responses to the target consonant (/d/); conversely, a high-F3 bias in context sounds produces more low-F3 responses (/g/). SCEs have been demonstrated using a variety of approaches, but most often, the context was a single sentence filtered two ways (*e.g.*, low-F3 bias, high-F3 bias) to introduce long-term spectral properties that biased speech categorization. Here, consonant categorization (/d/-/g/) was examined following context sentences that naturally possessed desired long-term spectral properties without any filtering. Filtered sentences with equivalent spectral peaks were included as controls. For filtered context sentences, as average spectral peak magnitudes (*i.e.*, filter gain) increased, SCE magnitudes (*i.e.,* category boundary shifts) increased linearly. However, unfiltered context sentences showed no relationship between long-term average spectral energy and subsequent SCEs. Instead, spectral energy in the last 500 ms of unfiltered sentences predicted SCE magnitudes. Results highlight important considerations about how long-term versus short-term spectral energy in preceding sounds affects categorization of later sounds.

4:30

4pPPc4. Complex frequency modulation detection and discrimination at low and high frequencies. Kelly L. Whiteford and Andrew J. Oxenham (Psych., Univ. of Minnesota, 75 East River Parkway, Minneapolis, MN 55455, whit1945@umn.edu)

Whether frequency modulation (FM) is represented by place (tonotopic) or temporal (phase-locking) information in the peripheral system may depend on the carrier frequency (f_c) and the modulation rate (f_m), with only f_cs below 4 kHz and f_ms below 10 Hz thought to involve temporal coding. This study tested the role of temporal coding in harmonic complexes by measuring FM detection and discrimination for two F0s (200 Hz and 1400 Hz), modulated at slow (1 Hz) and fast (20 Hz) rates, for tones with lower (2-5) or upper (6-9) harmonics embedded in threshold equalizing noise. Pure-tone FM detection was measured for f_cs between 200 and 12000 Hz at the same f_ms. In detection tasks, participants selected which of two intervals contained FM. In discrimination tasks, participants selected which of three FM complex tones was incoherently modulated. Preliminary results suggest better FM detection for slow than fast rates, even when all tones are above 4 kHz. However, this effect was stronger at lower frequencies, where phase-locking cues may be viable. Most participants were only able to discriminate inharmonicity for slow f_ms in conditions with components below 8000 Hz. [Work supped by NIH grant R01DC005216 and an Eva O. Miller Fellowship.]

4:45

4pPPc5. Frequency modulation improves sensitivity to interaural timing differences at high frequencies in the presence of noise. Alan Kan (Univ. of Wisconsin-Madison, 1500 Highland Ave, Madison, WI 53705, ahkan@waisman.wisc.edu)

Sensitivity to interaural timing differences (ITDs) at high frequencies has largely been attributed to slow amplitude modulations (AM) in the envelope. However, in noisy situations, high-frequency signal envelopes can easily be corrupted making AM less useful for sound localization. Previous work has shown that frequency modulation (FM) introduced into a high frequency pure tone can also yield sensitivity to ITDs. This suggests that FM may be an additional cue used by the auditory system to resolve high-frequency ITDs, which may be useful in noisy situations. This experiment tests the hypothesis that in-phase AM and FM are complementary cues that improve ITD sensitivity at high frequencies in the presence of noise. A 4-kHz tone stimulus with a fixed ITD of 500 μs was used, and had either 200 Hz AM, FM or in-phase AM+FM imposed. The signal-to-noise ratio (SNR) needed to achieve 70.7% correct in a left-right discrimination task was measured. Results showed that the SNR needed for FM was typically lower than AM, and AM+FM was significantly lower than either AM or FM alone. The results suggest that in-phase AM and FM may provide complementary cues that aid in discriminating ITDs at high frequencies in the presence of noise.

5:00

4pPPc6. The gammawarp filterbank: A frequency-warped implementation of the dynamic compressive gammachirp filter. James Kates (Speech Lang. Hearing Sci., Univ. of Colorado, 409 UCB, Boulder, CO 80309, James.Kates@colorado.edu)

This paper presents an alternative method for implementing a dynamic compressive gammachirp (dcGC) auditory filterbank. Instead of using a cascade of second-order sections, the new approach uses digital frequency warping to give the gammawarp filterbank. In the warped implementation, the unit delays in a conventional finite impulse-response (FIR) filter are replaced by first-order allpass filters. The set of warped FIR filter coefficients is constrained to be symmetrical, which results in a filter phase response that is identical for all filters in the filterbank. The dynamic variation in filter response is provided by interpolating the outputs of three linear filters adjusted for low-, medium-, and high-input signal levels. The gammawarp filterbank offers a substantial improvement in execution speed compared to previous dynamic gammachirp filter implementations. For a linear filterbank, the gammawarp execution time is comparable to that of a gammatone or linear second-order section gammachirp implementation for 32 bands, but there is a processing time advantage that increases as the number of bands is increased. For a dynamic compressive gammachirp filterbank, the gammawarp implementation is 27 to 47 times faster than the dcGC MATLAB code of Irino.

5:15

4pPPc7. Musical instrument categorization is highly sensitive to spectral properties of earlier sounds. Ashley Assgari, Jonathan Frazier, and Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, 317 Life Sci. Bldg., Louisville, KY 40292, ashley.assgari@louisville.edu)

Auditory perception is shaped by spectral properties of surrounding sounds. For example, when spectral properties differ between earlier (context) and later (target) sounds, this can produce *spectral contrast effects* (*SCEs*; *i.e.*, categorization boundary shifts) that bias perception of later sounds. SCEs influence perception of speech and nonspeech sounds alike. When categorizing vowels or consonants, SCE magnitudes increased linearly with greater spectral differences between contexts and target speech sounds [Stilp *et al.* (2015) *JASA*; Stilp & Alexander (2016) *POMA*; Stilp & Assgari (2017) *JASA*]. Here, we tested whether this linear relationship between context spectra and SCEs generalizes to nonspeech categorization. Listeners categorized musical instrument targets that varied from French horn to tenor saxophone. Before each target, listeners heard a one-second music sample processed by spectral envelope difference filters that amplified / attenuated frequencies to reflect the difference between horn and saxophone spectra. By varying filter gain, filters reflected part of (25%, 50%, 75%) or the full (100%) difference between instrument spectra. As filter gains increased to reflect more of the difference between instrument spectra, SCE magnitudes increased linearly, parallel to speech categorization. Thus, a close relationship between context spectra and biases in target categorization appears to be fundamental to auditory perception.

4p THU. PM

Session 4pSA

Structural Acoustics and Vibration, Physical Acoustics, and Engineering 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

1:30

4pSA1. Bianisotropic acoustic metasurfaces for wavefront manipulation. Benjamin M. Goldsberry, Andrew J. Lawrence, and Michael R. Haberman (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd, Austin, TX 78758, haberman@arlut.utexas.edu)

Materials with sub-wavelength asymmetry and long-range order have recently been shown to respond to acoustic waves in a manner that is analogous to electromagnetic biansiotropy [Sieck *et al*. Phys. Rev. B **96**, 104303, (2017)]. A characteristic of these materials is the generation of dipolar scattering when subjected to a uniform time-varying pressure field. This behavior leads to a characteristic acoustic plane-wave impedance that has a preferential direction, known as a polarization, which results in direction-dependent magnitude and phase of an acoustic field scattered from bianisotropic acoustic media. These materials can therefore be used as acoustic metasurfaces to control the reflected or transmitted acoustic fields. This work presents the use of bianisotropic acoustic media for wavefront manipulation through analytical and finite element modeling and discusses potential acoustic analogues to existing bianisotropic electromagnetic metasurfaces. [This work was supported by ONR and the National Science Foundation.]

1:50

4pSA2. Wave control with "Time Materials". Mathias Fink (Langevin Inst., 1 rue Jussieu, Paris 75005, France, mathias.fink@espci.fr)

Phononic crystals and Metamaterials are made from assemblies of multiple elements usually arranged in repeating patterns at scales of the order or smaller than the wavelengths of the phenomena they influence. Because time and space play a similar role in wave propagation, wave propagation is affected by spatial modulation or by time modulation of the refractive index. Here we emphasize the role of time modulation. We show that sudden changes of the medium properties generate instant wave sources that emerge instantaneously from the entire wavefield and can be used to control wavefield and to revisit the way to create time-reversed waves. Experimental demonstrations of this approach will be presented More sophisticated time manipulations can also be studied in order to extend the concept of phononic crystals in the time domain.

Contributed Papers

2:10

4pSA3. Reciprocity theorems and perception of non-reciprocal behavior in vibrating systems, imetamaterials, acoustic devices, and electroacoustic devices. Allan D. Pierce (Cape Cod Inst. for Sci. and Eng., PO Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537, allanpierce@verizon.net)

Equations governing physical systems often admit a reciprocity corollary. Such systems include mechanical, acoustical, vibrational, and electroacoustic systems. There is no universally applicable succinct definition of reciprocity, although it is sometimes loosely stated that reciprocity means that, "if you see it, then it can see you." There have been a considerable number of derivations of reciprocity theorems for different types of acoustical and mechanical systems published over the years — notable names are Helmholtz, Rayleigh, Lamb, Lyamshev, Schottky, Foldy, Primakoff, Payton, de Hoop, Knopoff, Gangi, Arntsen, Carcione, and Fokkema. There appears to be no universally applicable derivation, and no universally applicable reciprocity principle. The emergence of metamaterial systems poses new challenges and opportunities in regard to reciprocity. Given one such system, one naturally asks if there is some reciprocity theorem that applies. If one perceives that there is no such theorem, then one might naturally ask whether the perception is wrong, and if it is simply the case that no one has yet bothered to try to derive one or that one is not sufficiently clever to find one. It may also be that there is an analogous system for which there is a theorem, and that the theorem applies for the present system in some approximate sense. If so, how well? Present paper examines several metamaterial systems introduced in the recent literature from the viewpoint point of reciprocity theorems.

2:25

4pSA4. Nonlinear properties of an electrically coupled acoustic metamaterial. Jason J. Smoker and Alexey S. Titovich (Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd, West Bethesda, MD 20817, jason.smoker@navy.mil)

Efforts in the field of acoustic metamaterials have often remained in the material design and property identification domain due to difficulty in realizing physical manifestations of necessary material properties. This is especially true in the domain of transformation acoustics. In a number of classical approaches, research has relied on physical dynamics in the linear region to design an effective density or bulk modulus. Such work has limited achievable designs. This work aims to demonstrate the use of electrically coupled non-linear dynamics to increase the design space of acoustic metamaterials. Numerical models are leveraged for analysis and utilized to demonstrate efficiencies in real world configurations.

2:40

4pSA5. Extreme anisotropy over an acoustic hyperbolic metasurface. Li Quan and Andrea Alu (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, 1616 Guadalupe St., UTA 7.215, Austin, TX 78712, liquan@utexas.edu)

The emergence of optical hyperbolic metamaterials has attracted significant attention in the last decade due to its extreme anisotropy, linked to largely enhanced local density of states. This concept has been also extended to acoustics, and acoustic hyperbolic metamaterials have been realized. However, the fabrication complexity of these metamaterials, their inherent sensitivity to loss and disorder, and the fact that the hyperbolic modes are not easily accessible, as they are trapped in the bulk of the material prevent the wide use of hyperbolic metamaterials for sound. Our research group has recently proposed the idea of hyperbolic metasurfaces, for which extreme anisotropy over a surface enables plasmons that propagate with similar unbounded local density of states, but restricted to surface propagation. In this work, we extend these findings to acoustics, simplifying the realization of hyperbolic propagation and observation, and at the same time preserving the exciting properties of hyperbolic metamaterials. By carefully designing the acoustic microstructure decorating a metasurface, we successfully confine acoustic waves traveling along the interface with air and observe hyperbolic dispersion and in principle unbounded local density of states. These properties may open exciting opportunities for acoustic and surface acoustic wave devices.

2:55

4pSA6. Non-reciprocal bilinear structures. Andrew Norris, Zhaocheng Lu (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, norris@rutgers.edu), Samuel P. Wallen, Benjamin M. Goldsberry, Michael R. Haberman (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), Tyler J. Wiest, and Carolyn C. Seepersad (Mech. Eng., Univ. of Texas, Austin, TX)

Structures that display non-reciprocal behavior resulting from nonlinear effects are presented. The nonlinear mechanism is bilinear stiffness, also known as bimodular elastic response. Specific realizations considered are in the form of bilinear springs in series making a finite degree of freedom unidimensional nonlinear structure. Recall that a system is reciprocal if F_B $u_{B,A}$= F_A $u_{A,B}$ where is $u_{B,A}$(respectively $u_{A,B}$) is the displacement at point *B* (*A*) resulting from a force F_A applied at *A* (F_B applied at *B*). We say the system is fully non-reciprocal if the reciprocity relationship is violated for any sign of the applied forcing, positive or negative, compressive or tensile. Perhaps the simplest system displaying full non-reciprocity is a two degree-of-freedom spring-mass-spring-mass-spring structure, fixed at both ends. We first describe the static and low frequency behavior, illustrating full non-reciprocity. Similar systems with many bilinear springs and masses display nonlinear traveling wave effects, including pulse spreading and shock formation depending on whether the leading edge of the incident pulse is compressive or tensile, respectively. The talk will also discuss fabrication and optimization of the bilinear springs using additive manufacturing. Work supported by NSF-EFRI.

3:10–3:25 Break

3:25

4pSA7. Linear nonreciprocal transmission of sound in a viscous fluid with asymmetric scattteres. Arkadii Krokhin (Phys., Univ. of North Texas, 1155 Union Circle # 311427, Denton, TX 76203, arkady@unt.edu), Ezekiel Walker (Echonovus Inc, Denton, TX), Andrii Bozhko, and Arup Neogi (Phys., Univ. of North Texas, Denton, TX)

Reciprocity theorem is hold for wave equation due to time-reversal symmetry. Dissipative losses introduced by adding imaginary part to elastic modulus make sound propagation irreversible but the reciprocity theorem remains valid. In viscous fluid acoustic oscillations of local velocity follow the Navier-Stokes equation, which is not time-reversible. However, broken T-symmetry does not necessarily lead to nonreciprocity. Here we demonstrate that the necessary condition for nonreciprocal propagation in a viscous fluid is broken P-symmetry, which can be achieved by introducing asymmetric scatterers in a viscous environment. The experiment was done with a phononic crystal of Al rods in water. The rods have asymmetric cross-section in a form of a circular sector with an arc of 120 degrees. The measured transmission spectrum exhibits signatures of nonreciprocity within a range of frequencies from 300 to 450 kHz. The experimental results are in agreement with numerical simulations based on the linearized Navier-Stokes equation. The nonrecipocity is due to different viscous losses accumulated along sound propagation in two opposite directions with broken P-symmetry. Nonreciprocity increases for the rods with more rough surfaces since dissipation occurs mainly in a viscous boundary layer of thickness of few microns. Unlike previously proposed nonreciprocal devices based on nonlinearity or local spinning of fluid, our passive device is small, cheap, and does not require energy source. [Work supported by the NSF under Grant No. 1741677.]

3:40

4pSA8. Optimal absorption associated with wall impedance in acoustic waveguides. Matthew Kelsten and Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, mjk308@scarletmail.rutgers.edu)

In a previous talk at ASA it was shown that the double root for modal frequencies in a 2D waveguide can exhibit almost perfect absorption over a relative broad frequency range. The key to the phenomenon is that the wall impedance is such that modes coalesce at a complex-valued frequency. In this talk we consider the more realistic case of circularly symmetric waveguides. Numerical simulations are shown for reflection and transmission from a waveguide with impedance boundary conditions associated with double roots. The results again show large and almost perfect absorption at frequencies near the real part of the complex root. The model is linked to realistic passive impedance values based on various models for boundary impedance, including perforated and porous panels. These comparisons are discussed to illustrate the feasibility of optimized wall impedances in absorbing sound passing through ducts. Preliminary experimental data will be presented. Work supported by NSF.

3:55

4pSA9. Tunable acoustic energy dissipation by fluid-structure interaction in elastomeric, resonant metamaterials. 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)

Helmholtz resonators are a classic means to absorb low frequency acoustic waves with great effectiveness and straightforward implementation. Yet, traditional Helmholtz resonators are tuned to absorb a specific frequency of wave energy and are unable to adaptively tailor damping capability. On the other hand, recent research on elastomeric metamaterials offers concepts for large and tunable damping properties using internal constituents that buckle to trap large elastic energy when subjected to geometric or load constraints. The research reported here draws from the principles of Helmholtz resonance and constrained metamaterials to device a resonant metamaterial with tunable acoustic energy dissipation. Using the fluid-structure interaction of an internal beam-like member with the resonator chamber, the metamaterial absorbs acoustic waves at a target frequency and tailors energy dissipation by changing external constraints that relatively magnify or suppress the interaction between acoustic pressure and damped

beam member. An analytical model is developed to qualitatively characterize the behavior of the metamaterial observed in the laboratory. From the combined experimental and analytical studies, it is found that the metamaterial may significantly reduce the sound pressure level at the targeted frequency range while modulating the broadness of the absorption effect by way of external load control.

4:10

4pSA10. Acoustic gradient index lens design using gradient based optimization. Feruza Amirkulova, Samuel T. Caton, Morgan Schrader, Trenton J. Gobel (Mech. Eng., Western New England Univ., 1215 Wilbraham Rd., Springfield, MA 01119, feruza.amirkulova@wne.edu), and Andrew Norris (Mech. engineeringand Aerosp., Rutgers Univ., Piscataway, NJ)

We demonstrate a gradient based optimization of the absolute acoustic pressure at a focal point by incrementally repositioning a set of cylindrical obstacles so that they eventually act as an effective gradient index (GRIN) lens. The gradient-based optimization algorithm maximizes the absolute pressure at the focal point by analytically evaluating its derivative with respect to the cylinder positions and then perturbatively optimizing the position of each cylinder in the GRIN lens device while taking into account acoustic multiple scattering between the cylinders. The method is presented by examples of sets of hard cylinders and sets of elastic cylindrical shells of uniform size. Designs using available cylindrical shells enable easy and realistic fabrication of acoustic GRIN lenses. Computations are performed on MATLAB using parallel optimization algorithms and a multistart optimization solver, and supplying the gradients of absolute pressure at the focal point with respect to position vectors. Providing the analytic form of the gradients enhances modeling by allowing the use of exact gradients with optimization algorithms and parallel computing. This results in reducing number of function calls and time needed to converge, and improving solution accuracy for large scale optimization problems especially at high frequencies and with a large number of scatterers.

4:25

4pSA11. Distorting an impulse wave with phononic metamaterials– A numerical study. Jason R. Dorvee (US Army ERDC, US Army Corps of Engineers, ERDC, 72 Lyme Rd, Hanover, NH 03755, Jason.R.Dorvee@usace.army.mil), Michelle E. Swearingen (US Army ERDC, Champaign, IL), Donald G. Albert, Michael Muhlestein (US Army ERDC, Hanover, NH), Megan A. Krieger (US Army ERDC, Champaign, IL), and James L. O'Daniel (US Army ERDC, Vicksburg, MS)

This numerical study is focused on the effects of disordered arrays as phononic metamaterials on the distortion of low-frequency impulses. Through numerical simulations we have been able to determine the characteristic properties of such phononic metamaterials as they correlate to the disruption a propagating impulsive wave. The fast 2-D FDTD model used allows for rapid reconfiguration of theoretical designs of: arrangements, geometries, and materials for the meta-atom; the results are informing evolving theory of the disruption of impulsive waves using metamaterials. With the ability to share CAD designs with the computational model, the layouts of arrangements from the numerical simulations can be ported to a 3D printer for the fabrication of physical models, for validation of the numerical model. These simulations serve as a planning tool for the configuration of sensors for both bench-scale and field testing. The model used and results of the numerical study are presented. [The work described in this document was funded under the US Army Basic Research Program under PE 61102, Project T22, Task 01 "Military Engineering Basic Research"; Task 02 "Material Modeling for Force Protection;" Project T25, Task 01 "Environmental Science Basic Research" and was managed and executed at the US Army ERDC.]

4:40

4pSA12. Distorting an impulse wave with phononic metamaterials—An experimental study. Michelle E. Swearingen (US Army ERDC, Construction Eng. Res. Laborartory, P.O. Box 9005, Champaign, IL 61826, michelle.e.swearingen@usace.army.mil), Jason R. Dorvee, Donald G. Albert, Michael Muhlestein (US Army ERDC, Hanover, NH), Megan A. Krieger (US Army ERDC, Champaign, IL), and James L. O'Daniel (US Army ERDC, Vicksburg, MS)

Bench-scale experiments were performed to characterize impulsive signal distortion with metamaterial arrays. As physical models, the composition of these systems rely on acoustically rigid materials in order to be examined at these small scales. In the final design the propagation of the pulse was evaluated using an array of receivers. Unique elements of this experimental methodology include: the design and refinement of an anechoic chamber for high frequencies, an in-lab tunable impulse source for both high frequencies and high amplitude, bench scale experimental designs achieved through the 3-D printed configurations of computer aided designs (CAD) extracted directly from a numerical model for exacting representations of configurations, and a measurement scheme designed to map the 3-D distortion of the impulse wave. Data and a description of the experiment and results are presented. [The work described in this document was funded under the US Army Basic Research Program under PE 61102, Project T22, Task 01 "Military Engineering Basic Research"; Task 02 "Material Modeling for Force Protection;" Project T25, Task 01 "Environmental Science Basic Research" and was managed and executed at the US Army ERDC.]

Session 4pSC

Speech Communication: Non-Native Speech Perception and Production (Poster Session)

Jessamyn L. Schertz, Chair
Centre for French and Linguistics, Dept. of Lang. Studies, University of Toronto Scarborough, Suite 301, Erindale Hall, 3359 Mississauga Rd., Mississauga, ON L5L1C6, Canada

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

Contributed Papers

4pSC1. The influence of talker accent strength on accent categorization. Taylor Federspill and Tessa Bent (Speech and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, tfedersp@indiana.edu)

Listeners' identification accuracy for nonnative-accented words tends to decrease as the talker's accent strength increases. However, the influence of accent strength on accent categorization has not been investigated. We hypothesized that nonnative talkers with stronger accents would have more salient cues to their native languages and thus, listeners would show increased accent categorization accuracy. To test this hypothesis, monolingual American English listeners categorized sentences produced by 12 native and 24 nonnative talkers in a 12-alternative forced-choice task with 8 nonnative accent categories and 4 regional dialect categories. Three talkers from each language/dialect background with different accent strengths (i.e., mild, moderate, strong) were included. The talkers' accent/dialect strengths were determined by 20 listeners' ratings on a 9-point scale (nonnative talkers: 1 = no foreign accent and 9 = very strong foreign accent; native talkers: 1 = Standard American English and 9 = very strong regional dialect). For the nonnative talkers, preliminary results show a moderate correlation between accent categorization accuracy and accent strength. However, within specific language backgrounds, adherence to the expected accuracy pattern varied substantially. Future analyses will investigate how accent strength influences confusion patterns across accents and dialects. [Work supported by an Indiana University Hutton Honors College grant.]

4pSC2. Acoustic correlates of L2 Spanish judgments of accentedness and comprehensibility: A mixed-effects modeling approach. Ziwei Zhou and Charles Nagle (Iowa State Univ., 2603 Kent Ave. # 209, Ames, IA 50010, ziweizh@iastate.edu)

Two constructs—comprehensibility and accentedness—figure themselves prominently in listeners' judgments of L2 (second language) speech. Correlational analyses have shown that they make separate contributions to such judgments (Derwing and Munro, 1995, 1997, 2005). As it is important to set realistic goals for adult L2 learners by prioritizing understanding over nativelikeness (Levis, 2005), recent studies began to examine the relative contribution of linguistic aspects, especially acoustic features, to comprehensibility and accentedness (Munro & Derwing, 1999; Kang, Rubin, & Pickering, 2010; Trofimovich & Issac, 2012). Consistent with this agenda, this study first examines the reliability of scores (produced by the 9-point Lickert scale) under modern measurement framework (e.g. multi-facet Rasch measurement model). Then, the relationship between acoustic features, including the confidence values (for each utterance) obtained using Google Cloud Platform speech recognition engine as well as suprasegmental features extracted using Prosogram (Mertens, 2014), and ratings are established using mixed-effects modeling techniques. Results indicate that, even though it can be shown that comprehensibility and accentedness are statistically distinct constructs, both ratings share a set of common acoustic correlates—speech time, standard deviation of pitch values, and normalized pairwise variability index (nPVI).

4pSC3. Word recognition for regional dialects and nonnative accents in children and adults: Influence of psycholinguistic distance. Rachael F. Holt (Speech and Hearing Sci., Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210, holt.339@osu.edu), Tessa Bent (Speech and Hearing Sci., Indiana Univ., Bloomington, IN), Miller E. Katherine, Mone Skratt Henry, Melissa M. Martin, Izabela A. Jamsek, and Donna E. Green (Speech and Hearing Sci., Ohio State Univ., Columbus, OH)

Few studies have directly compared adults' or children's perception of nonnative accents and unfamiliar regional dialects. However, some evidence suggests that nonnative varieties cause greater decrements in intelligibility and processing than unfamiliar native dialects, while metalinguistic awareness for nonnative varieties develops earlier than awareness for regional variants. To directly examine regional and nonnative accent perception, we tested sentence recognition in American-English monolingual 5- to 7-year-old children and adults for three accents: Central Midland (familiar native), Scottish (unfamiliar native), and German (detectable but mild nonnative accent) in quiet and multitalker babble. In quiet, both children and adults showed highly accurate word recognition for all accents. Although children's performance was lower than adults' in noise, overall word recognition accuracy patterns across accents were similar: accuracy was highest for the Midland talker, followed by the German-accented talker, and poorest for the Scottish talker. These data suggest that the greater decrements for nonnative accents compared to unfamiliar regional dialects previously reported may have arisen from the specific varieties or talkers selected. Although both types of unfamiliar speech can cause listening difficulty in noisy environments, the acoustic-phonetic distance from the home dialect may predict both adults' and children's performance better than native vs. nonnative status.

4pSC4. The effect of clear speech on listener perception of nonnative speech. Sarah H. Ferguson (Dept. of Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Rm. 1201, Salt Lake City, UT 84112, sarah.ferguson@hsc.utah.edu), Alison Behrman (Speech-Language-Hearing Sci., Lehman College/City Univ. of New York, New Rochelle, NY), Jessica J. Staples, Stephanie M. Williams, and Paige E. Ihrig (Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT)

Listener perception of nonnative speech may be measured as intelligibility, ease of understanding, or accentedness. The relationship among these variables is complex, as is the influence of clear speech on these measures. The purpose of this study was to explore the effect on listener perception of clear speech produced by Spanish-accented talkers of American English.

Specifically, four questions guided this study: Does clear speech enhance intelligibility and ease of understanding? What is the effect of clear speech on accentedness? How does speaking style (conversational versus clear) influence the relationship among these perceptual variables? Does the presence or absence of supporting contextual information modify the relationship among intelligibility, ease of understanding, and accentedness? Twenty-eight female talkers of American English (half native English-speaking and half Spanish-accented) read aloud 56 short sentences using conversational and clear speech. The 56 sentences were comprised of 28 pairs, each containing a high- or low-probability final word. Young, normal-hearing, monolingual English listeners transcribed and rated ease of understanding for sentences presented in 12-talker babble, and then rated accentedness for sentences presented in quiet. Listeners assigned ratings using visual analog scales. Results and their implications for models of listener perception and clinical management of accentedness will be discussed.

4pSC5. Sentence recognition in noise: The interplay between talker intelligibility, familiarity, and linguistic complexity. Dorina Strori (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, dorina.strori@northwestern.edu), Ann Bradlow (Linguist, Northwestern Univ., Evanston, IL), and Pamela Souza (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

Foreign-accented speech recognition is typically tested with linguistically simple sentences, which offer a limited window into speech processing. In this study, participants transcribed simple (i.e. mono-clausal, canonical declarative syntax) and complex (i.e. multi-clausal, non-canonical syntax, and/or passive voice) sentences in noise at two signal-to-noise ratios. The sentences were spoken by three talkers: one native, one high and one low intelligibility L2 English speaker. Unsurprisingly, recognition accuracy dropped with decreasing intelligibility and signal-to-noise ratio. Interestingly, accuracy dropped with increasing linguistic complexity only for the native and high intelligibility talkers. For the low intelligibility talker, sentence complexity did not affect recognition accuracy. This pattern of findings was present in mixed and blocked experimental designs, indicating no effect of familiarity with the talker. We argue that listeners employ qualitatively different speech processing strategies for low versus high intelligibility talkers, and suggest that this difference may be driven by the talker's realization of prosodic structure. While native and high intelligibility talkers effectively encode the contrast between complex and simple sentences, low intelligibility talkers may not convey sufficient prosodic information, resulting in the recognition of both sentence types as non-hierarchically organized word sequences. [Work supported by Knowles Hearing Center, NIH grants R01-DC012289 and R01-DC005794.]

4pSC6. Interference of adaptation to foreign-accented speech from consecutive but not interleaved training of two accents. Kali Woodruff Carr and Beverly Wright (Commun. Sci. & Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60201, KALI@U.NORTHWESTERN.EDU)

For perceptual learning on fine-grained discrimination tasks, improvement can be enhanced or disrupted when two tasks are trained, depending on how the training on those tasks is distributed. To investigate if this phenomenon extends to speech learning, we trained native-English speakers to transcribe both Mandarin- and Turkish-accented English sentences using one of three different configurations of the same training stimuli. After training, all trained groups performed better than untrained controls, and similarly to each other, on a novel talker of a trained accent (Mandarin). However, for a novel accent (Slovakian), performance was better than untrained controls when training alternated between the two accents, but not when the two accents were trained consecutively. Performance for the novel accent decreased as the number of contiguous sentences per accent during training increased. One interpretation of these results is that accent information is integrated during a restricted time window. If two accents are encountered within this window, information from both accents is integrated, yielding accent-general learning. If two accents are encountered in separate consecutive windows, accent-specific learning occurs for each accent and accent-general learning is prevented. These results mirror patterns for fine-grained discrimination learning, and illuminate the processes underlying the success of high-variability training.

4pSC7. The influence of lexical characteristics and talker accent on the recognition of English words by speakers of Korean. Minkyoung Choi (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), Jeffrey J. Holliday (Dept. of Korean Lang. and Lit., Korea Univ., Bloomington, IN), and Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu)

A growing body of evidence that the influence of phonological neighborhood density (PND) on speech perception increases over the course of second language (L2) learning (e.g., Bradlow *et al*., 1999; Imai *et al*., 2005; Yoneyama & Munson 2017). This was first shown by Imai *et al*. (2005), who found that only high-proficiency Spanish-speaking L2 English-language learners (ELL) showed an influence of English PND on speech perception; low-proficiency ELLs did not. More recently, Yoneyama and Munson (2017) examined whether Imai *et al*.'s findings could be replicated with ELLs whose L1 was Japanese. In contrast to Imai *et al*., Yoneyama and Munson found that there were strong effects of PND on word recognition for all three groups of subjects: low-proficiency ELLs (tested in Japan), high-proficiency ELLs (tested in the US), and native speakers of English. The contrast between Imai *et al*. and Yoneyama and Munson's findings suggests that a wider set of studies using many different L1-L2 mismatches are needed to understand the range of PND effects in L2 learning. The current study examines Korean-speaking L2 learners of English using the same methods as Imai *et al*. and Yoneyama and Munson. Testing is ongoing in the US and Korea.

4pSC8. Generalization in foreign accent adaptation is predicted by similarity of segmental errors across talkers. Jordan Hosier (Northwestern Univ., 2016 Sheridan Rd, Evanston, IL 60208, jhosier@u.northwestern.edu), Bozena Pajak (Duolingo, Inc., Pittsburgh, PA), Ann Bradlow, and Klinton Bicknell (Northwestern Univ., Evanston, IL)

Comprehension of foreign-accented speech improves with exposure. Previous work demonstrates that listeners who adapt to accented talkers generalize that adaptation to other accented talkers—exposure to multiple talkers of the same accent facilitates comprehension of a novel talker of that accent (e.g. Bradlow and Bent, 2008) and exposure to multiple novel accents facilitates comprehension of yet another novel accent (Baese-Berk *et al*., 2013). To examine possible theories of accent adaptation and generalization, we created a new dataset of phonetically transcribed accented speech produced by the training and test talkers used as stimuli in studies of accented speech generalization (Bradlow and Bent, 2008; Baese-Berk *et al*., 2013). Using this dataset, we computed the (cosine) similarities between accented talkers in a multidimensional accent space defined by the rates at which talkers made different segment-level phonetic errors. Results show that similarity in this space accounts for all significant differences between training conditions in Bradlow and Bent (2008) and Baese-Berk *et al*. (2013): training conditions including more of the same types of phonetic errors as those of the test talkers led to better test talker comprehension. These results suggest that prior accent generalization results are compatible with simple, segment-error driven theories of adaptation.

4pSC9. Perception of non-native clear speech: The gap between speakers' effort and actual increase in intelligibility. Misaki Kato and Melissa M. Baese-Berk (Linguist, Univ. of Oregon, 161 Straub Hall, 1290 University of Oregon, Eugene, OR 97403, misaki@uoregon.edu)

Non-native speakers are less effective at increasing intelligibility in clear speech than native speakers. However, examining only intelligibility may not fully address how non-native clear speech is perceived by listeners. We examined native English listeners' perception of non-native English speakers' clear speech in terms of intelligibility, comprehensibility, and speakers' effort. Non-native English speakers produced English sentences in conversational and clear speaking styles. These productions were evaluated by 117 native English listeners. In an intelligibility task, listeners typed what they heard. In two rating tasks, listeners heard the same sentence produced by the same speaker in both a conversational and a clear speaking style. One group of listeners were asked to choose the sentence that was easier to understand (comprehensibility). Another group of listeners were asked to choose the sentence that the speaker was trying to say more clearly (speakers' effort). The results showed that conversational-clear

modifications did not result in increased intelligibility or comprehensibility. However, the rating task showed that listeners were sensitive to the speakers' effort in the clear speaking style. These results suggest that non-native speakers' effort to speak clearly may not necessarily result in intelligibility gains, but listeners may still be sensitive to this effort.

4pSC10. Cognitive effort in comprehending non-native speech decreases with listening experience. Dave C. Ogden (Linguist, Univ. of Michigan, 440 Lorch Hall, 611 Tappan Ave., Ann Arbor, MI 48109, ogdend@umich.edu)

Listeners adapt to non-native accents, as shown by improved accuracy transcribing a speaker's intended utterance during testing (Baese-Berk *et al*., 2013; Bradlow & Bent, 2008; Sidaras *et al*., 2009). However, this improvement does not address whether cognitive effort decreases with accent experience, or whether accented speech continues to require effort to compensate for atypical features. The present study addresses this question using pupil dilation, a physiological measure indexing cognitive effort (Sirois & Brisson, 2014). In a matched-guise experiment, 48 participants heard 90 sentences spoken in either a General American Accent (GAA) or a non-native accent (NNA) while an eye tracker measured pupil diameter. An equal number of sentences with words predictable from context (e.g., "Sugar tastes very sweet.") and without (e.g., "It can't be brown.") were included. Participants transcribed (typed) each sentence. Pupil dilation while participants listened to each sentence was measured as the difference between average diameter for 100 ms immediately before and after the sentence played. Dilation decreased significantly more in the NNA than in the GAA condition, indicating that perception of accented speech becomes less effortful with experience. No effects of semantic predictability were observed. The relationship between effort and transcription accuracy will be discussed.

4pSC11. Vowel perception in quiet and noise for native and non-native listeners: Analysis of vowel identifiability. Chang Liu (Commun. Sci. and Disord., The Univ. of Texas at Austin, 1 University Station A1100, Austin, TX 78712, changliu@utexas.edu), Su-Hyun Jin (California State Univ. at Northridge, Northridge, CA), and Sha Tao (Beijing Normal Univ., Beijing, China)

For a number of years, research on phonetic perception has been primarily focused on the percentage correct of phonetic identification (e.g., hit rate) without considering listeners' response (e.g., false alarm). The purpose of this study was to compute d' of English vowel identification in quiet and noise including both hit rate and false alarm for native and non-native listeners. Both data analyses showed native listeners outperformed non-native listeners for vowel identification in quiet and noise. However, the d' data showed that the non-naïve disadvantage became larger in long-term speech shaped (LTSS) noise than in quiet, while the percentage correct data showed an oppositely finding that the non-native disadvantage was smaller in LTSS noise than in quiet. These preliminary results suggest that false alarms should be included in the data analysis of phonetic perception and the idenitifiability (e.g., d') based on the signal detection theory may be more appropriate to evaluate listeners' capacity to identify phonemes. [Work supported by China Natural Science Foundation 31628009 and University of Texas at Austin Research Grant].

4pSC12. Speech perception for native and non-native English speakers: Effects of contextual cues. Ling Zhong, Chang Liu (Commun. Sci. and Disorder, Univ. of Texas at Austin, 2504A Whitis Ave. Stop A1100 Austin, TX 78712, Austin, TX 78712, ling.zhong@utexas.edu), Cuicui Wang, and Sha Tao (Beijing Normal Univ., Beijing, China)

Previous studies have shown that native English speakers outperformed non-native English speakers in perceiving English speech under quiet and noisy listening conditions. The purpose of this study is to investigate the difference between native English speakers and native Chinese speakers on using contextual cues to perceive speech in quiet and multi-talker babble. Three types of sentences served as the speech stimuli in this study: sentences with high predictability (including both semantic cues and syntactic cues), sentences with low predictability (including syntactic cues), and sentences with zero predictability (consisting random sequences of words). These sentences were presented to native-English and native-Chinese listeners in quiet and four-talker babble with the signal-to-noise ratio at 0 and -5 dB. Preliminary results suggested that native Chinese speakers primarily rely on semantic information when perceiving speech in quiet, whereas native English speakers showed greater reliance on syntactic cues when perceiving speech in noisy situations. The difference between native English speaker and native Chinese speakers on syntactic, and semantic information utilization in various listening conditions will be discussed. [Work supported by The University of Texas at Austin, Undergraduate Research Fellowship and China National Natural Science Foundation 31628009]

4pSC13. Identifying /l/ and /n/ in English by multi-dialectal speakers of Mandarin. Bin Li (Linguist and Translation, City Univ. of Hong Kong, 83 Tat Chee Ave., Kowloon Tong 000, Hong Kong, binli2@cityu.edu.hk)

Native speakers of Standard Mandarin are mostly proficient in their local dialects which were acquired at home before or parallel to the Standard Mandarin. The Mandarin dialects that share general grammar mainly differ in their phonological features. A well-known example is the contrast of /l/ and /n/, which is not fully maintained or completely absent in many southern Mandarin dialects. This study investigated how perception of the English consonantal contrast may be influenced by experience in Mandarin dialects with varying phonemic status and distribution of the sounds. We examined the identification of the two consonants in various phonetic environments by Chinese bi-dialectal speakers who were all advanced learners of English-as-a-foreign-language. Results confirmed predicted perceptual difficulties across dialect groups as well as revealed other phonetic factors affecting the perceptual accuracy.

4pSC14. Speaker variability in spoken word recognition: Evidence from non-native listeners. Yu Zhang (Commun. Sci. and Disord., Oklahoma State Univ., 48 E Stimson Ave Apt 2, Athens, OH 45701, yu.zhang10@okstate.edu)

Non-native listeners face many challenges during spoken word recognition. On the lexical level, the vocabulary size of non-native listeners is usually reduced relative to native listeners. Lexical competition can arise from the lexicon in both the native and the non-native language during speech perception. How do non-native listeners resolve speaker variability, a challenge that is presumably non-lexical in spoken word recognition? This study aims to explore non-native spoken word recognition in the face of speaker variability. Mandarin speakers using English as a second or foreign language are recruited to listen to pairs of English words (e.g., cat—dog). Lexical decision time and accuracy on the second item in a pair (e.g., dog) are measured. The pairs are either spoken by the same or different native speakers of American English to examine the effect of speaker change on reaction time and accuracy in a short-term priming paradigm. Non-native listeners' pattern of response is explored as a function of the amount of priming received against speaker variability.

4pSC15. Speech learning in noise or quiet. Lin Mi (State Key Lab. of Cognit. Neurosci. and Learning and IDG/McGovern Inst. for Brain Res., Beijing Normal Univ., Dept. of Psychiatry, The Affiliated Brain Hospital of Guangzhou Medical University (Guangzhou Huiai Hospital), 36 Mingxin Rd., Liwan District, Guangzhou The Affiliated Brain Hospital of Guangzhou Medical University (Guangzhou Huiai, China, linmi83@hotmail.com), Sha Tao, Wenjing Wang, Qi Dong (State Key Lab. of Cognit. Neurosci. and Learning and IDG/McGovern Inst. for Brain Res., Beijing Normal Univ., Beijing, China), and Chang Liu (Dept. of Commun. Sci. and Disord., the Univ. of Texas at Austin, Austin, TX)

Noise made speech perception much more challenging for non-native listeners than for native listeners. We proposed that speech training in noise may better improve non-native listeners'speech perception. In the present study, three groups of native Chinese listeners were trained on English vowel perception under three conditions including quiet (TIQ), noise (TIN), and watching videos in English as an active control (C). Vowel identification in quiet and noise was assessed before, right after training, and three months later after training. Results showed that all listeners significantly

improved their vowel identification after training, and the training effect was retained three months later. Compared with the control group, the TIQ group improved more for vowel identification in quiet condition, but not for that in noise conditions. The TIN group improved vowel identification in noise significantly more than the TIQ and control groups. Compared with the TIQ, less informational masking and energetic masking was also found in the post-training test for the TIN than that for the TIQ. These results suggested that vowel perceptual training in noise may better improve L2 vowel perception in adverse acoustic environments than training in quiet. Implications for second language speech perception training were discussed.

4pSC16. Voice onset time and onset f0 as correlates of voicing in American learners of French. Amy Hutchinson and Olga Dmitrieva (Purdue Univ., 100 N. University St., West Lafayette, IN, hutchi25@purdue.edu)

Voice Onset Time (VOT) and onset f0 are known correlates of voicing distinctions in stops and both contribute to the perception of voicing (House & Fairbanks, 1953; Abramson & Lisker, 1965). The values of VOT and onset f0 which correspond to voicing categories vary cross-linguistically. Second language (L2) learners often have to acquire a novel use of these acoustic cues to produce and perceive L2 voicing. The acquisition of primary voicing cue, VOT, has been studied extensively in L2 research (Flege & Eefting, 1988; Flege 1991; Birdsong *et al.* 2007) but less is known about the acquisition of secondary cues. The present study compares the use of VOT and onset f0 in French and English speech produced by American learners of French (22). The study also examines the role of back transfer in L2 learners by comparing their English productions to a monolingual control group (33). The results demonstrate that although learners' VOT values in French were heavily influenced by English, their onset f0 production in both English and French was on target. Little evidence of learners' second language affecting their first language was found. Individual trends, including the effect of L2 proficiency level will also be explored.

4pSC17. First language phonetic drift in second language instructional environment. Olga Dmitrieva and Alexis N. Tews (Purdue Univ., 640 Oval Dr., Stanley Coulter 166, West Lafayette, IN 47907, odmitrie@purdue.edu)

Recent research demonstrates that second language learning can affect first language speech production. This has been shown for proficient long-term second language learners in immigration settings (Flege, 1987) as well as novice learners in the study abroad setting (Chang, 2013). The question addressed in the present study is whether learners acquiring a second language in a classroom setting in the first language-dominant environment are also subject to L1 phonetic drift. A group of American students enrolled in intermediate-level Russian language courses at a major Mid-Western university were recorded reading a list of Russian and English words designed to investigate the acoustic realization if word-initial and word-final stop voicing. Several acoustic correlates of initial and final voicing were measured (VOT, onset f0, preceding vowel duration, glottal pulsing, etc.) and compared to data from monolingual speakers of American English from the same geographic region. Results demonstrate that learners' realization of both initial and final voicing in English are affected by exposure to Russian. In the majority of measures, the drift occurred in the direction of norms characteristic of Russian voicing. Divergence from Russian was detected in the use of negative VOT for initial voiced stops.

4pSC18. The effect of distributional training on the production of non-native vowel contrasts. Grechen Go, Heather Campbell, and Susannah V. Levi (Communicative Sci. and Disord., New York Univ., 665 Broadway, 9th Fl., New York, NY 10012, gretchen.go@nyu.edu)

The current study tests production accuracy of a non-native vowel contrast following a modified perceptual learning paradigm. Previous research has found that adults can learn non-native sound categories with sensitivity to distributional properties of their input. In the current study, 34 native-English adults heard stimuli from /o/-/oe/ continuum. Half of the participants heard stimuli drawn from a bimodal distribution and half from a unimodal distribution. To support learning, we incorporated active learning with feedback, lexical support in the form of two images, and overnight consolidation. Production of this contrast was assessed through a repetition task at baseline and at the end of the experiment. Production accuracy is measured as the Euclidean distance between /o/ and /oe/ at baseline and post-training. Preliminary analyses suggest that the distance between these two vowels increased for both groups, indicating that listeners in both conditions were able to transfer perceptual learning to production. These results suggest a way to mitigate the disadvantages previously found for participants in the unimodal condition, by incorporating active engagement with the target stimuli using lexical support, accuracy feedback, and overnight consolidation.

4pSC19. Disruption of learning on a non-native phonetic contrast by a brief practice break. David F. Little (Elec. and Comput. Eng., Johns Hopkins Univ., 1316 Eutaw Pl., #31, Baltimore, MD 21217, david.frank.little@gmail.com) and Beverly Wright (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

What is required for long-lasting learning of a non-native phonetic contrast? Do principles from fine-grained discrimination learning apply? Discrimination learning often requires extensive practice within a day for performance to improve across days, and a 30-minute break midway through practice can disrupt that learning. Thus it appears that trials must integrate to a learning threshold in a transient memory store in order for durable discrimination learning to occur. We asked whether this same principle applies in phonetic-contrast learning. To do so, we trained monolingual speakers on a non-native phonetic contrast along a voice-onset-time (VOT) continuum. Those who engaged in a single continuous practice session on day 1 reached above chance accuracy on day 2 while classifying the non-native phoneme. In contrast, when day-1 practice included a 30-minute break halfway through training, day-2 performance was at chance. These results suggest that for phoneme learning, as for discrimination learning, long-lasting learning requires that practice trials be integrated up to a learning threshold within a transient memory store before they are sent en masse into a memory that lasts across days.

4pSC20. Effect of L1 phonation contrast on production of L2 English stops. Jessamyn L. Schertz (Dept. of Lang. Studies, Univ. of Toronto Mississauga, Ste. 301, Erindale Hall, 3359 Mississauga Rd., Mississauga, ON L5L1C6, Canada, jessamyn.schertz@utoronto.ca)

This work examines the influence of native phonation contrast type on the production of English stops in different phonetic contexts/reading styles. Proficient English speakers from four different L1 language backgrounds produced words in three different contexts: words in isolation, phrase-final in a carrier sentence, and in a reading passage. Language backgrounds were representative of three types of phonation contrasts: Mandarin (aspiration contrast: [p^h ~ p]), Tagalog (voicing contrast: [p ~ b]), and Urdu (4-way phonation contrast [p^h ~ p ~ b^h ~ b]). 11 participants from each group were compared with a control group of L1 English speakers. Aspiration and closure voicing were measured. All groups produced English voiceless stops as aspirated; however, there was considerable variation in the voiced stops, with Mandarin speakers producing less voicing, and Urdu and Tagalog speakers producing more voicing, than English speakers across speech styles, showing an asymmetrical influence of L1 phonation on production of the English contrast, in line with other recent work.

4pSC21. The production of American English vowels by native Japanese speakers in two different conditions. Takeshi Nozawa (Lang. Education Ctr., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, t-nozawa@ec.ritsumei.ac.jp)

Native Japanese speakers produced American English vowels /i, ɪ, ɛ, æ, ɑ, ʌ/ in two different conditions. First they read aloud one syllable words and non-words that contain these vowels on the word list. The vowels produced in this condition are believed to be what Japanese speakers believe typical of each English vowel. The Japanese speakers also reproduced English vowels uttered by native speakers. They were not told what vowel they would hear in each trial. The vowels produced in this way should be close to how each vowel sounds to the Japanese speakers, regardless of whether they correctly identify each vowel. The results revealed that in the "read aloud" conditions, /

i/ and /ɪ/ are spectrally close, and so are /æ/ and /ʌ/ and in low central position. /ɑ/ is higher than /æ/ and /ʌ/. In "reproduction" condition, on the other hand, /i/ and /ɪ/ are spectrally more distant, and /æ/ is fronted. /ɑ/ and /ʌ/ spectrally closer than in "read aloud" condition. Taken together, it can be assumed that there is a discrepancy between how Japanese speakers believe English vowels sound and how English vowels actually sound to them.

4pSC22. Cross-modality interference effects of phonetic and prosodic information in second language learners. Chieh Kao and Yang Zhang (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr SE, Minneapolis, MN 55455, kaoxx096@umn.edu)

Successful communication relies on efficient use of information from multiple modalities and dimensions. Our previous research using a cross-modal priming paradigm showed that congruent visual primes facilitate faster recognition of phonetic and emotional prosodic information in speech. Event-related potential (ERP) data further revealed distinct brain mechanisms in the N400 and late positive response (LPR) components for processing linguistic and paralinguistic congruency. The current study extended the same paradigm to English-as-a-second-language (ESL) learners to examine possible interference between the two informational dimensions as a function of language experience. Participants were sixteen normal adult ESL learners. Monosyllables /bab/ and /bib/ in a happy or angry tone were used as the auditory stimuli, and pictures of the speaker articulating vowel /a/ and /i/ with a happy or angry facial expression were used as the visual primes. Compared to native English speakers, ESL learners showed significantly longer reaction time with inconsistent congruency effects in both conditions. But their behavioral accuracy data mirrored those of the native speakers, and native-like N400 and LPR components were also reliably elicited in the ESL group in both conditions. Together, these results indicate stronger cross-modal interference between phonetic and emotional prosodic information in speech for second language learners.

4pSC23. Second-language learning in adolescents with cochlear implants. Deniz Baskent, Dorit Enja Jung (Dept. of Otorhinolaryngology, Univ. Medical Ctr. Groningen, Univ. of Groningen, PO Box 30.001, Groningen 9700RB, Netherlands, d.baskent@umcg.nl), Wander Lowie (Appl. Linguist, Ctr. for Lang. and Cognition, Univ. of Groningen, Groningen, Netherlands), and Anastasios Sarampalis (Psych., Univ. of Groningen, Faculty of Behavioural and Social Sci., Groningen, Netherlands)

Speech signals delivered via cochlear implants (CIs) lack spectro-temporal details, yet, young-implanted children can develop good native language skills (L1). This study explores three research questions: 1. Can adolescents with CIs learn a second language (L2)? 2. Is there a difference in spoken (auditory-A) vs. written (visual-V) L2 skills? 3. Which perceptual and cognitive factors influence L2 learning? Two groups (L1 = Dutch, age 12—17 years), one with normal hearing (NH) and one with CIs, and both learning English (L2) at school, participated. L1 and L2 proficiency was measured in receptive vocabulary (A), comprehension (A, V), and general proficiency (V). Further, basic auditory functioning, in temporal (gap detection) and spectral (spectral ripple detection) resolution, and cognitive functioning, in IQ, working memory, and attention, were measured. Preliminary data ($n = 7$ per group) indicated comparable L1 proficiency between NH and CI groups. While some CI users showed L2 proficiency within the NH range, on average, L2 proficiency was lower for the CI group. This effect was more pronounced for auditory tests. Reduced temporal and spectral resolution, but no difference in cognitive tests, were observed in CI group compared to NH, emphasizing the importance of auditory factors in L2 learning.

4pSC24. Production and perception of phonetically "non-native" clusters by Georgian native speakers. Harim Kwon (George Mason Univ., 5 rue Thomas Mann, CEDEX 13, Paris 75205, France, harim.kwon@univ-paris-diderot.fr) and Ioana Chitoran (Université Paris Diderot, Paris, France)

This study examines the role of native inter-consonant timing patterns in perception and production of word-initial consonant clusters, asking how phonotactically native clusters with non-native timing patterns are perceived and produced by speakers of a cluster-heavy language. We tested Georgian speakers in two experiments using CCV/CVCV stimuli produced by a French talker. Georgian is a cluster-heavy language with a relatively long inter-consonant lag, often resulting in transitional vowels between two consonants within a cluster. French onset clusters have shorter inter-consonant lag than Georgian ones. In Experiment 1 (shadowing), Georgian participants (n = 14) were exposed to French CCV/CVCV stimuli and asked to produce what they heard. In Experiment 2 (discrimination), the same Georgian participants heard the same French stimuli in AX pairs and determined whether the two sequences (A and X) were the same or different. The results from the two experiments suggest that Georgian participants often confused French CCV (without any vocalic transition) with French CVCV. The confusion occurred almost exclusively when the first vowel of CVCV was /ø/, which is acoustically similar to the transitional vowel in Georgian, but did not seem to stem from Georgian speakers' incapability of producing French short-lag CCV. We claim that native inter-consonant timing patterns influence perception and production of consonant clusters even when the clusters are phonotactically licit in one's native language.

4pSC25. Does it have to be correct?: The effect of uninformative feedback on non-native phone discrimination. Annie J. Olmstead and Navin Viswanathan (Speech, Lang., Hearing, Univ. of Kansas, 1000 Sunnyside Ave, Lawrences, KS 66045, annie.j.olmstead@ku.edu)

Learning the phonetic inventory of a non-native language requires perceptual adjustment to non-native phones that sometimes belong to a single category in the learner's native language. For example, English native speakers often struggle to learn the distinction between the Hindi phonemes [ʈ] and [t̪] that are both categorized as [t] in American English (AE). Olmstead and Viswanathan (2017) showed that AE listener's discrimination of these non-native phones could be improved using short exposure to naturally produced Indian English (IE) words that contained the target contrast. In the current study, we examine how feedback affects this lexical retuning effect. Specifically, we set up feedback schedules that either reinforce the consistent mapping of these consonants onto the AE speaker's existing [t] and [θ], or that reinforce an inconsistent mapping. If the consistency of this mapping in IE is paramount to improving phonetic discrimination, then reinforcing it should strengthen the effect and providing a variable mapping should weaken it. Results suggest that the feedback given to participants did not change the effect on discrimination—discrimination improved whether or not the feedback was consistent. Implications of these findings are discussed.

4pSC26. Using high variability phonetic training to train non-tonal listeners with no musical background to perceive lexical tones. Alif Silpachai (Appl. Linguist and Technol., Iowa State Univ., 527 Farmhouse Ln., Ames, IA 50011, alif@iastate.edu)

Previous research has not extensively investigated whether High Variability Phonetic Training (HVPT) is effective in training listeners with no musical background and no prior experience with a tone language in their identification of non-native lexical tones. In this study, it was investigated whether HVPT is applicable to the acquisition of non-native tones by such listeners. Twenty-one speakers of American English were trained in eight sessions using the HVPT approach to identify Mandarin tones in monosyllabic words. Ten of the participants were exposed to words produced by multiple talkers (MT condition), and eleven participants were exposed to words produced by a single talker (ST condition). The listeners' identification accuracy revealed an average 44% increase from the pretest to the posttest for the MT condition and an average 30% increase for the ST condition. The improvement also generalized to new monosyllabic words produced by a familiar talker and those produced by two unfamiliar talkers. The learning however did not generalize to novel disyllabic words produced either by a familiar talker or an unfamiliar talker. Comparisons between two groups further revealed that the improvement of the listeners in the MT condition was significantly higher than that of the listeners in the ST condition.

4pSC27. Perceiving allophonic alternations in an immersion setting: The case of alveolars and rhotics in Spanish and English. Farrah A. Neumann (Linguist, Univ. of Pittsburgh, 4200 5th Ave, Pittsburgh, PA 15260, fan9@pitt.edu)

Recent research indicates that second language (L2) immersion influences the phonetic production not only of the L2 (Jacobs, Fricke, & Kroll, 2016), but also of the first language (L1) (Chang, 2012). However, less understood is the immediate impact of L2 immersion on cross-language influence in phonetic *perception*. This study tracked L2 Spanish study abroad learners' perceptual shifts due to immersion using auditory-visual cross-modal priming experiments in both Spanish and English. The target phones, [r, ɾ, t, d, ð, ɹ], were chosen for their differing phonemic and allophonic statuses in Spanish and English. For example, Spanish rhotics include the phonemic trilled /r/ and tapped /ɾ/, whereas the tapped /ɾ/ exists phonetically in English, but as an allophone of the alveolar stops /t d/. Because learners must adjust their phonological inventories to accommodate the L2 statuses of these phones, we expect their L1 inventories to also reflect these changes, as evidenced by slower RTs for the prime-target pairs that do not correspond to the L2. This study provides a fuller perspective on how immersion mediates L1-L2 interference, and offers insight as to whether perceptual changes in the two languages are necessarily related. Data analyses are currently underway.

4pSC28. Language experience outweighs perceptual enhancement in category learning. Meng Yang (Linguist, Univ. of California, Los Angeles, 3125 Campbell Hall, Los Angeles, CA 90095, mengyang@ucla.edu)

Some theorists claim that cues that contribute to the same acoustic percept are naturally coupled in category learning. In contrast, others claim that no pairs of cues are privileged in this way, and listeners simply learn cue correlations from language input. We compared the cue weighting and cue-shifting between pitch and breathiness, which are enhancing but do not signal a phonemic contrast in English (Experiment I), and pitch and closure duration, which are not enhancing but are correlated in signaling the English voicing contrast (Experiment II). When the cues were enhancing but not contrastive, English listeners successfully learned to weight the more informative cue higher. In contrast, preliminary results show that listeners have difficulty weighting the informative cue higher when the two cues are not enhancing but only contrastive. Results suggest that language experience outweighs auditory enhancement relationships in the perceptual coupling of cues.

THURSDAY AFTERNOON, 10 MAY 2018 GREENWAY H/I, 1:00 P.M. TO 4:30 P.M.

Session 4pSP

Signal Processing in Acoustics: Topics in Signal Processing in Acoustics

Ryan L. Harne, Cochair
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Buye Xue, Cochair
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Contributed Papers

1:00

4pSP1. Analysis of foldable acoustic arrays from piecewise linear, conformal, and tessellated topologies. 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)

Arrays of acoustic sources distributed over the tessellated frameworks of origami-inspired folding patterns demonstrate exceptional and straightforward adaptivity of acoustic wave guiding architecture to contrast to the challenges manifest in digital wave guiding control methods. Yet, when the concept of folding arrays is applied to tessellations formed by single patterns, acoustic fields with distinct distributions of energy delivery cannot be realized. To overcome this limitation, here multiple folding patterns are assembled by piecewise geometries that permit the array to conform to unique topologies when reconfigured by physical folding actions. An analytical model is developed to predict the acoustic pressure radiated from the acoustic sources arranged on piecewise-continuous surfaces, while experiments are conducted to demonstrate the concept and validate the model. Parametric investigations are then undertaken to uncover the relations of modularity and folding of the acoustic array and the resulting acoustic energy delivery and distributions. The results find that the assembly of tessellated arrays from multiple distinct folding pattern sub-elements may replicate the acoustic wave guiding of complex acoustic sources that traditionally do not enable adaptive wave radiation properties.

1:15

4pSP2. Cross-correlation method for acoustic detection of small unmanned aerial vehicles. Alexander Sedunov, Hady Salloum, Alexander Sutin, and Nikolay Sedunov (Maritime Security Ctr., Stevens Inst. of Technol., One Castle Point on Hudson, Stevens Inst. of Technol., Hoboken, NJ 07030, asedunov@stevens.edu)

The availability of Unmanned Aerial System (UAS) to consumers has increased in the recent years, with it came the potential for negligent or nefarious misuse of them. Stevens Institute of Technology has built a passive acoustic system for low flying aircraft detection, the application of the developed principles and algorithms for UAS acoustic detection and tracking is presented in this paper. The application of the developed principles and algorithms for UAS acoustic detection and tracking is presented in this paper. Several experiments were conducted aiming to establish the characteristics of the emitted noise of UAVs of various sizes while airborne and demonstrate the processing required to detect and find the direction toward the source. The vehicles tested included popular quadrotors: DJI Phantom 2 Vision+, 3DR Solo, DJI Inspire 1 as well as larger semi-professional vehicles: Freefly Alta 6, DJI S1000. The small array of 16 microphones was used for data collection in the tests near local NJ airport. Acoustic signatures of the tested UAS were collected for stationary and flying UAS. We applied the algorithm for detection and direction finding based on fusing time difference of arrival (TDOA) estimates computed by finding peaks in the output

Generalized Cross-Correlation (GCC) function. The cross-correlation signal process provided UAS detection and bearing for distances up to 250m while the spectrograms did not reveal acoustic UAS signatures at that distance. This work is being supported by DHS's S&T Directorate.

1:30

4pSP3. Cross-term analysis in frequency-difference-based source localization methods. Brian M. Worthmann (Appl. Phys., Univ. of Michigan, 1231 Beal Ave, 2010 Automotive Lab, Ann Arbor, MI 48104, bworthma@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

In previous work, it has been shown that a quadratic product of frequency-domain acoustic fields at different frequencies but the same spatial location leads to an auxiliary field which may contain field information at frequencies below the original signal's bandwidth (Worthmann, Song and Dowling, 2015, JASA 138, 3549-3562). This quadratic product, termed the frequency-difference autoproduct, has been shown to be valuable for beamforming and source localization in the presence of environmental mismatch and/or array sparseness. However, in a multipath environment, this quadratic product leads to undesired cross-terms. Bandwidth averaging procedures have been found to suppress some of their detrimental influences in some cases, but not all. Additionally, the poor dynamic range observed in frequency-difference beamforming and frequency-difference matched field processing are associated with the imperfect mitigation of these cross-terms. In this presentation, the nature of these cross-terms is analyzed, and signal processing tools are developed which attempt to robustly mitigate the detrimental effects of these cross terms. These signal processing tools can be used to potentially improve localization performance when using frequency-difference autoproduct-based source localization schemes. [Sponsored by ONR and NSF]

1:45

4pSP4. Investigation of reconfigurable antennas by foldable, e-textile tessellations: Modeling and experimentation. Chengzhe Zou (Mech. and Aerosp. Eng., The Ohio State Univ., Columbus, OH), Shreyas Chaudhari, Saad Alharbi, Hamil Shah, Asimina Kiourti (Elec. and Comput. Eng., The Ohio State Univ., Columbus, OH), and Ryan L. Harne (Mech. and Aerosp. Eng., The Ohio State Univ., 201 W 19th Ave, E540 Scott Lab, Columbus, OH 43210, harne.3@osu.edu)

Physical deformation mechanisms are emerging as compelling and simple ways to adapt wave propagation properties of antenna arrays in contrast to digital steering approaches acting on topologically fixed antennas. Concepts of physical reconfigurability also enable exceptional capabilities such as deployable and morphing antenna arrays that serve multiple functions and permit compact transport with ease. Yet, the emergent concepts lack broad understanding of effective approaches to integrate conformal, electrically conductive architectures with high-compliance foldable frameworks. To explore this essential interface where electrical demands and mechanical requirements may conflict, this research studies e-textile-based reconfigurable antenna arrays that conform to adaptable topologies by origami-inspired tessellations. The e-textiles leverage embroidered conductive threads along frameworks established on origami tessellations to permit large compliance at folding edges as needed while retaining the desired electromagnetic wave propagation characteristics. Computational modeling is used to guide experimental fabrications and validations. It is found that e-textile origami antenna arrays permit significant adaptation of wave radiation while maintaining the required mechanical robustness under folding sequences. These findings may motivate future concepts for reconfigurable antennas established upon physical deformation processes.

2:00

4pSP5. Mathematical properties of generalized Bessel functions and application to multi-tone sinusoidal frequency modulation. Parker Kuklinski (Naval Undersea Warfare Ctr., 1176 Howell St, Newport, RI 02841, parker.s.kuklinski1@navy.mil) and David A. Hague (Naval Undersea Warfare Ctr., North Dartmouth, Massachusetts)

The generalized Bessel function (GBF) is a multi-dimensional extension of the standard Bessel function. Two dimensional GBFs have been studied extensively in the literature and have found application in laser physics, crystallography, and electromagnetics. However, a more rigorous treatment of higher-dimensional GBFs is lacking. The GBF exhibits a rich array mathematical structure in regards to its partial differential equation representation, its asymptotic characterization, and its level sets. In this talk/paper, we explore these properties and connect spectral and ambiguity function optimization of a multi-tone SFM signal to finding the location of the roots of these generalized Bessel functions.

2:15

4pSP6. Consensus detection of a Gaussian underwater acoustic source by a distributed sensor network. Thomas J. Hayward and Steven I. Finette (Naval Res. Lab., 4555 Overlook Ave SW, Washington, DC 20375, thomas.hayward@nrl.navy.mil)

Probabilistic data fusion is a well-established algorithmic approach to the detection of underwater acoustic sources by a distributed sensor network. Direct communication of all of the network nodes with a fusion center would provide optimal joint detection via summation of log-likelihood ratios (LLR's) across the nodes. However, a network that relies on a single fusion center for data exfiltration is not robust to the loss of the fusion center. Distributed detection mediated by dynamic consensus algorithms, which have their roots in the broader system-theoretic discipline of dynamic state estimation, offers an alternative that is robust to node loss. The present work illustrates the application of dynamic consensus algorithms to the distributed detection of underwater acoustic sources, taking as an example the detection of a stationary, ergodic Gaussian source in a shallow-water waveguide by a network of vertical-array nodes. A general definition of consensus algorithms is followed by the construction of particular linear consensus dynamical systems that (1) provide asymptotic agreement of the time-averaged consensus LLR with the optimal central-fusion LLR; and (2) equalize the mean cross-node disagreements of the consensus LLR's. Performance of centralized and consensus detection is compared in simulations. [Work supported by ONR.]

2:30–2:45 Break

2:45

4pSP7. Integration of signal processing modules of hearing aids as real-time smartphone apps. Nasser Kehtarnavaz and Linda Thibodeau (Univ. of Texas at Dallas, 800 West Campbell Rd., Richardson, TX 75080, kehtar@utdallas.edu)

Our research group has been working on developing an open source research platform for hearing improvement studies based on smartphones noting that currently no open source, programmable, and mobile research platform exists in the public domain for carrying out hearing improvement studies. Such a platform allows smartphones, which are already widely possessed and carried by people, to be used for the development and field testing of existing or new hearing improvement signal processing algorithms. A number of smartphones apps have already been developed by our research group for specific or individual modules of the speech processing pipeline of a typical digital hearing aid. This work covers the integration of several individual signal processing modules into an integrated smartphone app. The steps taken in order to enable the individual modules to run together in real-time and with low audio latency will be presented and a demo of the integrated app running on both iOS and Android smartphones will be shown.

3:00

4pSP8. Stereo I/O framework for audio signal processing on android platforms. Abdullah Küçük, Yiya Hao (Elec. and Comput. Eng., The Univ. of Texas at Dallas, 17878 Preston Rd, APT 369, Dallas, TX 75252, axk166230@utdallas.edu), Anshuman Ganguly, and Issa M. Panahi (Elec. and Comput. Eng., The Univ. of Texas at Dallas, Richardson, TX)

Smartphones and tablets are attractive cost-effective solutions for implementing multi-channel audio signal processing algorithms. Modern smartphones and tablets have at least two microphones and requisite processing capabilities to perform real-time audio signal processing. However, it is still challenging to integrate the dual-microphone audio capture and the audio

processing pipeline in real-time for efficient operation. In this paper, we propose an efficient stereo input/output framework for real-time audio signal processing on Android smartphones/tablets. This framework enables us to perform a large variety of real-time signal-processing applications on Android platform such as Direction of Arrival (DOA) estimation, Audio Compression, and Multi-Channel Speech Enhancement. The implementations and demos of Direction of Arrival (DOA), Audio Compression, and Multi-Channel Speech Enhancement algorithms are also presented to show the effectiveness of proposed framework.

3:15

4pSP9. Robust real-time implementation of adaptive feedback cancellation using noise injection algorithm on smartphone. Parth Mishra (Elec. and Comput. Eng., Univ. of Texas at Dallas, 7740 mccallum blvd, apt#315, DALLAS, TX 75252, pxm161830@utdallas.edu), Serkan Tokgoz, and Issa M. Panahi (Elec. and Comput. Eng., Univ. of Texas at Dallas, Richardson, TX)

We propose a novel low latency smartphone-based application that demonstrates the real-time operation to cancel the negative effects of acoustic feedback arising from the coupling between the speaker and the microphone of the smartphone or similar device utilizing the robust Noise Injection (NI) method. We make use of multiple noise injections of short durations to estimate the filter coefficients of an appropriate order between the speaker and the microphone, in order to perform the feedback cancellation effectively in real-time. Our motive behind the development of this application is to perform an effective acoustic feedback cancellation irrespective of the position of speaker and the microphone on the platform under consideration. With the proposed application, we can estimate the transfer function between speaker and microphone in the changing room acoustics making the feedback cancellation very effective. Objective and subjective tests were conducted and results of the proposed real-time application indicate significant acoustic feedback suppression in the presence of varying environmental conditions.

3:30

4pSP10. Comparison of processing speed between Arduino Due and Teensy 3.6 boards in a system identification application. Artur Zorzo (Acoust., UFSM, SQN 315 BLG APTO 205, ASA NORTE, BRASILIA 70774070, Brazil, arturzorzo@hotmail.com), William D. Fonseca, Paulo Mareze, and Eric Brandao (Acoust., UFSM, Santa Maria, RS, Brazil)

The system identification technique is a powerful real-time signal processing tool when dealing with the characterization of dynamic systems. This approach uses adaptive filtering techniques and the least mean square (LMS) algorithm to estimate the impulse response of a system in a finite impulse response (FIR) filter format. The main purpose of this paper is to compare the results and processing speed of two ARM-based microcontrollers as well as to give a theoretical background and an implementation procedure of the technique. ARM (Advanced RISC Machine) is an architecture of 32-bit processors usually applied in embedded systems due to its processing power, relatively low consumption, cost and small size. The impulse responses obtained by the method for several passive electronic filters were transformed to the frequency domain and compared with simulations of the circuits' frequency responses. This practice was repeated for both boards and for different parameters of the algorithm. The results have shown that the Arduino Due was unable to deal with higher sampling frequencies due to its processor limitations. On the other hand, the Teensy 3.6, a more powerful controller, yields great results even for higher frequencies.

3:45

4pSP11. Calibration of low-cost sound level meters using machine learning techniques. Jose Giraldo (AAC centro de acústica aplicada, circular 74b 39b-71, Medellin 050031, Colombia, jose091@gmail.com) and Alberto Bañuelos Irusta (AAC centro de acústica aplicada, Vitoria, Spain)

The assessment of environmental noise pollution from complex sources that are not represented in traditional noise maps requires a long term monitoring network to manage citizen complaints, a network of low-cost sound level meters is a practical option to analyze real-world cases, however the accuracy of the measurements is a concern when this information is used to evaluate regulation accomplishment. Previous work on low-cost monitoring devices has been carried out in microphone comparison, processing board selection and recently in MEMS microphone use. In the present work the use of reinforcement learning techniques is explored to calibrate the sound pressure levels generated by a low-cost monitoring device using a class one sound level meter as reference in a continuous measurement setup. Machine learning based models are known to take into account strong non-linearities which are effective to get an alternative calibration method for low-cost sensors.

4:00

4pSP12. Improved pre-filtering stages for GCC-based direction of arrival estimation using smartphone. Abdullah Küçük (Elec. and Comput. Eng., The Univ. of Texas at Dallas, 17878 Preston Rd, APT 369, Dallas, TX 75252, axk166230@utdallas.edu), Anshuman Ganguly, and Issa M. Panahi (Elec. and Comput. Eng., The Univ. of Texas at Dallas, Richardson, TX)

Sound source localization (SSL) is one of the important areas in signal processing especially for hearing aid applications. Having at least two microphones and powerful processing capability makes smartphone as a good sound source locator for people who have hearing impairment. In our previous work, we showed that traditional Generalized Cross Correlation (GCC) method with spatial post-filtering stage would increase the performance of instantaneous estimation of direction of arrival (DOA). In this paper, we combine a spectral pre-filtering stage with our previous work to improve the noise robustness of the earlier method. By combining pre-filtering stage with GCC and post-filtering stage, we obtain robust DOA estimation under various realistic noise types. We experiment with several pre-filtering stages and study their impact on reducing root mean square error (RMSE) and mean absolute error (MAE) for DOA estimation. The real-time implementation of some of the proposed algorithms on smartphone is also presented.

4:15

4pSP13. Speech enhancement and speaker verification with convolutional neural networks. Peter Guzewich and Stephen Zahorian (Binghamton Univ., PO Box 6000, Binghamton, NY 13902, zahorian@binghamton.edu)

Convolutional neural networks have been used with great success for image processing tasks. The convolution filters can act as powerful feature detectors in images to find edges and shapes. Time domain speech signals are one dimensional, but frequency domain information can be viewed in the same way as an image, allowing one to use two-dimensional convolutional networks for tasks such as speech enhancement. In this paper, recently reported deep neural network preprocessing methods used for dereverberation are examined to explore their ability to remove noise and or reverberation. A convolutional network is then introduced and compared to the previous methods to explore the relative strengths and weaknesses of each approach. Comparisons are made through experimentation in the context of speech quality and speaker identification tasks.

Session 4pUW

Underwater Acoustics: Underwater Acoustic Communications, Positioning, and Signal Processing

Aijun Song, Chair
Electrical and Computer Engineering, University of Alabama, 401 7th Ave, Hardaway Rm284, Tuscaloosa, AL 35487

Contributed Papers

1:00

4pUW1. Unsupervised seabed characterization with Bayesian modeling of SAS imagery. T. S. Brandes and Brett Ballard (BAE Systems, Inc., 4721 Emperor Blvd., ste 330, Durham, NC 27703, tsbrandes@gmail.com)

Seabed characterization has utility for numerous applications that seek to explore and interact with the seafloor, ranging from costal habitat monitoring and sub-bottom profiling to man-made object detection. In the work presented here, we characterize the seabed based on the texture patterns within SAS images constructed from high-frequency side-scan sonar. Features are measured from the SAS images (e.g., lacunarity, an established texture feature coding method, and a rotationally invariant histogram of oriented gradients). Based on these SAS image features, we perform unsupervised clustering with a hierarchical Bayesian model, which creates categories of seabed textures. Our clustering model is a new variant of the hierarchical Dirichlet process that is both adaptive to changes in the seabed and processes batches of SAS imagery in an online fashion. This allows the model to learn new seabed types as they are encountered and provides usable clustering results after each batch is processed, rather than having to wait for all the data to be collected before being processed. The model's performance of seabed characterization by SAS image texture is demonstrated in the overall range and internal consistency of textures specific to each learned cluster. [Work supported by the Office of Naval Research.]

1:15

4pUW2. Use of adaptive thresholding to improve accuracy of small-aperture acoustic localization in shallow water. Paul Murphy (Univ. of Washington, Mech. Eng., UW Mailbox 352600, Seattle, WA 98195-2600, pgmurphy@uw.edu) and Brian Polagye (Univ. of Washington, Seattke, WA)

Passive acoustic localizing arrays are utilized in shallow water for applications ranging from environmental monitoring to marine security. Although large-aperture hydrophone arrays can provide accurate estimates of source position, they can be costly to deploy and maintain. Small-aperture arrays can be used to estimate the incident azimuth and elevation angles of acoustic sources, but their effectiveness may be hindered by multipath interference when deployed in reverberant environments. Here, a small-aperture hydrophone array is used to estimate the direction of acoustic sources with center frequencies of 69 and 120 kHz and durations less than 5 ms which were deployed at ranges of up to 75 m in a bay with a water depth on the order of 10 m. Time-differences of arrival are determined using an adaptive thresholding algorithm, which is shown to have superior performance to a cross-correlation method due to the prevalence of multipath interference in the signals. Errors in source bearing estimates and in the underlying time-differences of arrival calculations are evaluated by comparing to ground-truth values generated from the GPS position of the source.

1:30

4pUW3. Time delay estimation via zero-crossing of envelope. Donghwan Jung and Jea Soo kim (Korea Maritime and Ocean Univ., 727 Taejong-ro, Yeongdo-Gu, Busan 49112, South Korea, wjdehdghkss@gmail.com)

Time-delay estimation of an acoustic signal in an ocean waveguide is a challenging due to the received signal corrupted by noise and phase distortion caused commonly by reflection of the transmitted signal traveling to a remote receiver via more than one path. In this study, envelope-based Hilbert transform method is applied to enhance the accuracy of time-delay estimation. Through the mathematical interpretation and numerical simulations, the proposed method is compared to the conventional time-delay estimation methods (e.g., cross correlation and zero-crossing of Hilbert transform). It is found that the envelope-based Hilbert transform method more efficiently estimates the time-delay than the conventional methods and resolve ambiguity of time delay estimate.

1:45

4pUW4. Tracking and classification of targets detected by a moving multibeam sonar. Emma D. Cotter (Mech. Eng., Univ. of Washington, 3900 E Stevens Way NE, Seattle, WA 98195, ecotter@uw.edu), James Joslin (Appl. Phys. Lab, Univ. of Washington, Seattle, WA), and Brian Polagye (Mech. Eng., Univ. of Washington, Seattke, WA)

There are multiple applications where tracking and classification of targets detected in multibeam sonar data is required, ranging from environmental studies to marine security. Target tracking from a fixed frame of reference is well-established, but if the sonar is mounted to a surface-following platform or vessel, the tracking task is complicated by the moving frame of reference. Here, accelerometer data is integrated into a Kalman filter-based tracking scheme to correct for the motion of a non-stationary sonar. This technique is first verified using a vessel-mounted sonar and target with known position. Tracking of marine animals from a moving platform is then demonstrated in real-time for the case of a sonar mounted on the hull of a wave energy converter. These target tracks are classified in real time (e.g., diving bird, seal) using a Random Forest algorithm. The effectiveness and limitations of real-time target tracking and classification from a moving multibeam sonar are discussed.

2:00

4pUW5. Time reversal based frequency-domain equalization for underwater acoustic single-carrier transmissions. Xingbin Tu, Aijun Song (Elec. and Comput. Eng., Univ. of Alabama, 401 7th Ave, Hardaway Rm-284, Tuscaloosa, AL 35487, song@eng.ua.edu), and Xiaomei Xu (Xiamen Univ., Xiamen, Fujian, China)

Time reversal processing generates a time-focused signal, after combining dispersive reception from a hydrophone array in the underwater acoustic

channel. Because of this property, the time reversal method has been studied as a preprocessing scheme to support decision feedback equalization for underwater acoustic communications. Orthogonal frequency division multiplexing has also been combined with time reversal processing to improve spectral efficiency, as the equivalent compact channel impulse response allows short prefixes in the acoustic channel. Here, we show that time reversal processing can be used to facilitate prefix-free frequency domain equalization. Traditional frequency domain equalization schemes rely on the use of periodic prefixes, which bring a loss of spectral efficiency in the already-bandlimited underwater channel. The core idea is to use time reversal preprocessing to generate a delta-like equivalent impulse response to facilitate robust reconstruction of prefixes at the receiver for prefix-free single-carrier transmissions. The resultant receiver not only improves the spectral efficiency, but also supports communications over channels of variable fluctuation rates. The effectiveness of the receiver has been tested using the field measurements from the Gulf of Mexico in August 2016.

2:15

4pUW6. M-ary cyclic shift keying spread spectrum underwater acoustic communication using frequency domain energy detector. zhuoqun wei, Xiao Han, and Jingwei Yin (College of Underwater Acoust. Eng., Harbin Eegineering Univ., 145 Nantong St., Harbin, Heilongjiang Province, Harbin, Heilongjiang 150001, China, weizhuoqun@hrbeu.edu.cn)

M-ary cyclic shift keying (M-CSK) method can greatly improve the communication rate of conventional direct sequence spread spectrum, but the robustness is seriously influenced by the fluctuation of sea surface and Doppler effect. As to this problem, M-ary cyclic shift keying spread spectrum (M-CSKSS) underwater acoustic communication (UAC) based on frequency domain energy detector (FDED) algorithm is proposed in this paper. Time reversal (TR) is firstly adopted to suppress multipath spread using channel impulse response (CIR) estimated from the former symbol. And then the baseband signal after TR processing goes through a FDED, decisions are made according to the output energy later. Simulation results show the robustness of this algorithm under low signal-to-noise ration and multipath environment. In a field experiment, data transmission with very low bit error rate (BER) was achieved and the communication rate was 375bits/s.

2:30–2:45 Break

2:45

4pUW7. A method of distinguishing between surface and underwater targets based on time delay in deep water. Chunpeng Zhao and Guolong Liang (Underwater Acoust. Eng., Harbin Eng. Univ., No.145 Nantong St.,Nangang District,Harbin,Heilongjiang Province,China, Harbin, Heilongjiang 150001, China, 1067283316@qq.com)

It is difficult to distinguish between surface and underwater targets in passive detecting with a single sensor or a horizontal linear array in deep water. It is noticed that targets in different depths have different characteristics in time delay difference of rays. This paper takes advantage of this feature to discriminate targets' depths in direct-arrival zone and shadow zone in deep water. And the relation between judgment boundary and depth of sensor is analyzed. The simulation result shows that when the depth of sensor is 50m, the boundary which can distinguish between surface and underwater targets can be set to 10m. And the depth of the boundary decreases with the increase of the depth of sensor.

3:00

4pUW8. Research on integrated positioning approach based on long/ultra-short baseline. Jingfei Wang, Nan Zou, and Fu Jin (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, Heilongjiang 150001, China, 527656530@qq.com)

With the development of underwater acoustic technology, the combination of multiple positioning means becomes the trend of underwater acoustic positioning technology. Based on an integrated long/ultra-short baseline location equipment, this paper discusses a multi-solution fusion method based on the minimum mean square error criterion. This method fuses long baseline and ultra-short baseline information on output, and it is relatively simple. The results of integrated long/ultra-short baseline location are compared with the results of long baseline location and ultra-short baseline location by simulation and experience. It shows that the integrated long/ultra-short baseline location system has the higher positioning accuracy than long baseline location and ultra-short baseline location. It realizes the high precision positioning of underwater target.

3:15

4pUW9. Study of long baseline optimization positioning algorithms Bbsed on redundancy measurement. Ying Guo, Yan Wang (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin 15000, China, guoyinggy@hrbeu.edu.cn), and Nan Zou (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, Heilongjiang, China)

Generally, underwater vehicles real-time navigation system works through the principle of spherical intersection. The number of array elements required for this method is fixed, but in the case of redundant array elements, they can't be fully utilized. This paper presents two optimization methods based on redundancy measurement. One approach is the steepest descent method and the other is Gauss-Newton method. The former searches for the minimum value of the measurement's sum of squared residuals along the negative gradient direction, and the later, a non-linear least-squares optimization method that constantly revise regression coefficients through multiple iterations to minimize the sum of squared residuals. Simulation show that both methods are greatly affected by the initial position, the closer the distance between initial position to the target is, the smaller positioning error is. On the same initial position, the Gauss-Newton method has less positioning error than the steepest descent method. The conclusions have been proved to be valid by sea trial.

3:30

4pUW10. Direction estimation of coherent signals based on the symmetry of uniform linear array. Yunqi Tian, Guolong Liang, and Fu Jin (College of Underwater Acoust. Eng., Harbin Eng. Univ., 145 Nantong St.,Harbin, Heilongjiang Province, Harbin, Heilongjiang 150001, China, tianyunqi@yahoo.com)

A large number of coherent signals are generated due to interface reflection in the ocean. A novel algorithm based on the symmetry of uniform linear array (ULA) is proposed to solve the problem of direction of arrival (DOA) estimation in coherent environment. Considering the center sensor as the reference, ULA can be divided into two symmetric parts. Firstly, the conjugate data of one part is rearranged, meanwhile, the other part data is maintained. Secondly, the correlation function calculated by processed data is averaged to construct a full rank Hermitian Toeplitz matrix. Finally, we can estimate the directions of coherent sources by eigen-decomposition. The results of simulation show that the proposed algorithm not only can effectively distinguish coherent signals, but also can improve the resolution, the accuracy and the probability of success in DOA estimation. Furthermore, the proposed algorithm is robust in the case of lower SNR or fewer snapshots.

3:45

4pUW11. An effective method of choosing vertical beam pitch angle of sonar array. Mengnan He, Fan Zhan, Guolong Liang, and Ao Shen (College of Underwater Acoust. Eng., Harbin Eng. Univ., No.145 Nantong St.,Nangang District, Harbin, Heilongjiang 150001, China, hmn111@126.com)

In order to obtain a better array gain, the sonar main-beam should point to the incident direction of acoustic waves. For stereoscopic sonars, such as cylindrical array sonar, the incident direction of acoustic wave contains both horizontal angle and vertical pitch angle (VPA). Due to the variation of acoustic VPA caused by hydrological condition, the array gain would decrease seriously when the VPA of sonar beam doesn't match the VPA of acoustic waves. To solve the above problem, we proposed a VPA optimization method for stereoscopic sonars. First,we analyzed the variation rule of both sonar array gain and the acoustic VPA. Then,search for the VPA of the sonar beam in the vertical directions by beam scanning, so as to maximize the array gain. Simulations results are provided to validate the performance of the proposed method.

4:00

4pUW12. Methods of suppressing tow ship noise with a horizontal linear array. Jia Feng, Nan Zou (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, Heilongjiang, China), Yan Wang (College of Underwater Acoust. Eng., Harbin Eng. Univ., Rm. 903, NO. 145, Nantong St., Harbin, China, wangyan@hrbeu.edu.cn), and Yu Hao (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, Heilongjiang, China)

Tow ship noise seriously influence the detection and direction of arrival (DOA) estimation performance of towed array. Null steering beamforming(NSBF), inverse beamforming(IBF) and non-data based beam-space transformation(NDBBT) are applied to suppress interference of tow ship in this paper. These methods are evaluated according to the improvement of signal interference noise ratio and DOA estimation precision. Simulation and experimental results show that all three methods can suppress interference of tow ship effectively so as to improve DOA estimation precision. In addition, IBF and NDBBT do better in suppressing interference than NSBF.

4:15

4pUW13. Generalized likelihood ratio test based on mode-space beamfoming with a horizontal line array. Yifeng Zhang (Underwater Acoust. College, Harbin Eng. Univ., Harbin NanGang District Nantong St. 145, Harbin, Hei Longjiang 150001, China, zhangyifeng2587@163.com), Nan Zou, Jinjin Wang, and Jia Feng (Underwater Acoust. College, Harbin Eng. Univ., Harbin, Heilongjiang, China)

The conventional detection method has difficulty in detecting the narrowband signal in the noisy shallow water waveguide, because detection under the plane-wave model mismatches the shallow water waveguide. In this paper, a generalized likelihood ratio test based on mode-space beamfoming is proposed, considering the acoustic field in a shallow water waveguide can be characterized by normal mode model. Due to the otherness of the mode wavenumbers, the generalized likelihood ratio test model is constructed by mode-space beamforming. As the mode-space beamfoming has an ill-posed problem, we ignore the small eigenvalues of the coefficient matrix, and then has a benefit on the performance of detector. Theoretical analysis and simulation result validate the effectiveness of the proposed algorithm.

THURSDAY EVENING, 10 MAY 2018 — 7:30 P.M. TO 9:00 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings. All meetings will begin at 7:30 p.m., except for Engineering Acoustics which will hold its meeting starting at 4:30 p.m.

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

Committees meeting on Thursday are as follows:

Noise	Nicollet D3
Speech Communication	Nicollet D2
Underwater Acoustics	Greenway B

Session 5aPA

Physical Acoustics: General Topics in Physical Acoustics II

Sam Wallen, Chair
The University of Texas at Austin

Contributed Papers

8:00

5aPA1. Generation of ultrasonic waves in vegetable tissue via non-ablative laser pulses. David A. Collazos-Burbano (Ctr. for Bioinformatics and Photonics (CIBioFi), Universidad del Valle, calle 13 No. 100-00, Univalle, 351, of 2009, Cali, Valle del Cauca 760032, Colombia, david.collazos@correounivalle.edu.co), Carlos A. Melo-Luna (Quantum Technologies, Information and Complexity Group (QuanTIC), Universidad del Valle, Cali, Valle del cauca, Colombia), Joao L. Ealo (Vibrations and Acoust. Lab. (LaVA), Mech. Eng. School, Universidad del Valle, Cali, Colombia), and John H. Reina (Quantum Technologies, Information and Complexity Group (QuanTIC), Universidad del Valle, Cali, Valle del Cauca, Colombia)

The study of vegetable tissue is essential in plant growth analysis. The elasticity is an important characteristic when firmness is studied in fruits or when turgidity is tested in plant leaves. This property gives information about the ripeness of fruits, the water content in leaves, and, in general, the morphological and physiological state of plants. Hence, the propagation of elastic waves in these media is a useful tool that gives information about the developmental stage of the plant. In this sense, laser ultrasonics is a promising method in the characterization of composite materials. In a certain way, vegetable tissues can be thought of as a kind of biological composite abstracted as a stratified plate of several layers. In this work, as a first approximation, we propose a methodology to generate guided waves in vegetable tissues by using laser ultrasonics techniques. We implement a pulsed laser system to induce ultrasonic waves without damage on the samples. Besides, we develop a methodology for processing the propagated acoustic waves. This analysis allows us the assessment of dispersive properties, a useful tool when determining important descriptors such as firmness index, water content, and provide useful information to feed biophysical models, among others.

8:15

5aPA2. Probing dynamic response to temperature changes in Berea Sandstone using acoustic time-of-flight and resonant ultrasound spectroscopy. Eric S. Davis, Cristian Pantea, Blake Sturtevant, and Dipen N. Sinha (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 and bulk resonance tracking were used to study the anomalous elastic behavior in Berea Sandstone with temperature cycling. Previously published results (Davis *et al*., 2016) using Resonant Ultrasound Spectroscopy (RUS) indicated that Berea Sandstone softens by as much as 10% when cooled from 205°C to 110°C. This result is in contrast to the behavior of most materials, which stiffen when cooled. The time-of-flight experiments confirmed the RUS results and revealed the need for additional analysis of this phenomenon. By tracking the location of resonant frequencies at each temperature step, the qualitative behavior of the sample was obtained without the need for a rigorous RUS analysis. The results from this analysis showed that Berea Sandstone has significant hysteretic behavior under temperature cycling. It was also revealed that the qualitative elastic behavior of Berea Sandstone is unchanged with increasing relaxation time from the thermal shock induced by rapid cooling of the sample.

8:30

5aPA3. Elastic constants of self-healing polyethylene co-methacrylic acid determined via resonant ultrasound spectroscopy. Kenneth A. Pestka II, Jacob W. Hull, Jonathan D. Buckley (Chemistry and Phys., Longwood Univ., 201 High St., Farmville, VA 23909, pestkaka@longwood.edu), and Stephen J. Kalista (The Dept. of Biomedical Eng., Rensselaer Polytechnic Inst., Troy, NY)

Self-healing polyethylene co-methacrylic acid (EMAA) is a thermoplastic material that can be shaped into structures that are capable of autonomously healing after being cut, punctured, or shot. In this work, we present results from several samples of self-healing EMAA-0.6 Na, with 60% of the methacrylic acid groups neutralized by sodium, known as DuPont Surlyn 8920. This ionomer, unlike typical crystals with extremely high quality factors and easily identifiable ultrasonic resonances, is a relatively soft polymeric material with significant damping, resulting in sample resonances that can overlap and are often difficult to identify and isolate. In this work, we show the resonant spectrum of several different EMAA-0.6 Na samples as well as the methods used to isolate individual resonances and determine the elastic constants.

8:45

5aPA4. An ultrafast broadband digital signal processing approach to pulse echo measurements. Blake Sturtevant, Dipen N. Sinha, and Cristian Pantea (Mater. Phys. and Applications, Los Alamos National Lab., MPA-11, MS D429, Los Alamos, NM 87545, bsturtev@lanl.gov)

Pulse-Echo (PE) measurements are commonly used to determine the sound velocity in a sample of known length. In PE measurements, the interface between the transducer and buffer rod, and that between the buffer rod and sample, introduces a total phase shift, Φ, to the acoustic wave that must be accounted for if high accuracy in the velocity is desired. The appropriate time correction, $\tau=\Phi/\omega$, is traditionally determined by measuring the time-of-flight of multiple acoustic waves with different angular frequencies, ω. A single PE measurement can take several minutes, depending on experimental details such as the number of frequencies measured and the amount of signal averaging performed. Several minutes of data acquisition is tolerable for static experiments, but it is too long for many dynamic processes of interest in biology, pulsed magnetic fields, or any other rapidly changing system. This work describes an approach to PE in which a single broadband signal is collected and later processed offline using standard digital signal processing techniques. We demonstrate that a single captured waveform can be processed to yield the same information as 80 separate waveforms collected over an 80 MHz span, enabling accurate PE measurements on sub-second time scales.

9:00

5aPA5. Two-point statistics of synthetic three-dimensional polycrystalline microstructures. Musa Norouzian and Joseph A. Turner (Dept. of Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, W342 Nebraska Hall, Lincoln, NE 68588-0526, jaturner@unl.edu)

The two-point spatial statistics of a heterogeneous material are crucial in ultrasonic attenuation and backscatter calculations. In this presentation, several common assumptions used for modeling polycrystals are examined. In most studies, the spatial and tensorial elements of the covariance function are assumed to be decoupled. Also, two-point correlation models often assume a single effective grain size for polycrystalline materials and a spatial correlation function that has an exponential decay at a rate directly proportional to the average grain diameter. This investigation uses the Dream.3D software platform to generate three-dimensional (3D) realizations of polycrystalline materials in order to study their two-point statistics. Using lognormal distributions with fixed means but different widths, a variety of different polycrystalline microstructures are simulated. The results show that the spatial correlation function is directly influenced by the grain size distribution. For a fixed mean grain size, wider distributions exhibit longer range correlations than narrower ones. The results also show that decoupling of the tensorial and spatial elements of the covariance function may not be valid for all lognormal grain size distributions. These results are anticipated to have an important impact on ultrasonic attenuation and backscatter models. [Research supported by AFRL under prime contract FA8650-15-D-5231.]

9:15–9:30 Break

9:30

5aPA6. Uncertainty bounds on ultrasonic phase velocities for polycrystalline media. Musa Norouzian and Joseph A. Turner (Dept. of Mech. & Mater. Eng., Univ. of Nebraska-Lincoln, W342 Nebraska Hall, Lincoln, NE 68588-0526, mnorouzian@huskers.unl.edu)

Most theoretical work related to the study of the effective properties of polycrystals assume infinite media with randomly oriented grains. Therefore, the bulk material has absolute isotropy. However, real samples always include a finite number of grains such that the inspection volume will have some associated anisotropy. Hence, bounds on the bulk properties can be expected for a given measurement. In this work, the effect of the number of grains on this anisotropy variation is studied using Dream.3D software. The effective elastic modulus tensor is derived using Voigt, Reuss, and self-consistent techniques with 1700 microstructures comprised of equiaxed cubic grains in 17 different volumes. The bond transformation is utilized to quantify the standard deviation of the average elastic modulus. The standard deviations of several materials are shown to be inversely proportional to the square root of the number of grains. Based on the single-crystal anisotropy, a master curve is derived which relates modulus anisotropy to the number of grains. In addition, Christoffel equation is used to study the relevant phase velocities. With appropriate normalization, a similar master curve is derived. Such results will have important impacts on ultrasonic models associated with metals. [Research supported by AFRL under prime contract FA8650-15-D-5231.]

9:45

5aPA7. A study on warning sound for drowsiness driving prevention system. Ik-Soo Ann (Cultural Contents, Soongsil Univ., 369 Sangdo-ro. Dongjak-gu, Seoul 156-743, South Korea, aisbestman@naver.com)

The Korea Expressway Corporation has studied various preventive measures to prevent drowsiness driving accidents, such as providing various drowsiness driven prevention measures, establishing drowsiness rest areas, and operating alarms on drowsiness driving in highway tunnel have been implemented. Recently, the Korea Transportation Safety has also used the monitoring of the driven state (DSM—Driven State Monitoring) and the driving information of the vehicle (VDI—Vehicle Driving Information) to confirm the state of the driver in the running car and perform a drowsiness driving and we are showing a good reaction by introducing a drowsiness driven prevention system which sounds vibration and warning sound. As described earlier, when a driver wakes up during a drowsiness operation, a warning sound plays an important role by installing it in a tunnel or in a car. In this paper, we further developed the warning sound for battle against drowsiness used in the tunnel of the existing highway and studied focusing on using inside the vehicle. This research aims to verify and further improve the warning sound part of the in-vehicle drowsiness driven prevention system which is just introduced and enforced, effectively and contribute effectively to prevent drowsiness driven accidents.

10:00

5aPA8. A study on the comparison of characteristics between blower and sound fire extinguisher. Seonggeon Bae (Div. of Comput. Media Information Eng., Kangnam Univ., 40, Kangnam-ro, Giheong-gu, Youngin-si, Gyeonggi-do, Youngin 446-702, South Korea, sgbae@kangnam.ac.kr) and Myungjin Bae (Information and Commun., Soongsil Univ., Seoul, South Korea)

In general, the characteristics of the blower used for cleaning and the characteristics of the sound fire extinguisher have the same point that they use wind. However, unlike a blower that allows the wind to clean by the wind, a sound extinguisher is a principle that turns off the fire by blocking the air of oxygen necessary for ignition in the air. This study was carried out by separating the two features.

10:15

5aPA9. A robust smartphone based multi-channel dynamic-range audio compressor for hearing aids. Yiya Hao (Engineer and Comput. Sci., Univ. of Texas at Dallas, 380 Vistacourt Dr., APT2317, Plano, TX 75074, yxh133130@utdallas.edu), Ziyan Zou (Engineer and Comput. Sci., Univ. of Texas at Dallas, Garland, TX), and Issa M. Panahi (Engineer and Comput. Sci., Univ. of Texas at Dallas, Richardson, TX)

Hearing impairment degrades perception of speech and audio signals due to low frequency-dependent audible threshold levels. Hearing aid devices (HADs) apply prescription gains and dynamic-range compressor for improving users' audibility without increasing the sound loudness to uncomfortable levels. Multi-channel dynamic-range compressor enhances quality and intelligibility of audio output by targeting each frequency band with different compression parameters such as compression ratio (CR), attack time (AT), and release time (RT). Increasing the number of compressor channels can result in more comfortable audio output when appropriate parameters are defined for each channel. However, the use of more channels increases computational complexity of the multi-channel compressor algorithm limiting its application to some HADs. In this paper, we propose a nine-channel dynamic-range compressor with an optimized structure capable of running on smartphones and other portable digital platforms in real time. Test results showing the performance of proposed method are presented too.

Session 5aPP

Psychological and Physiological Acoustics: Examining the Auditory Brain

Ross K. Maddox, Chair
Biomedical Engineering and Neuroscience, University of Rochester, 601 Elmwood Avenue, Box 603, Room 5.7425, Rochester, NY 14642

Contributed Papers

8:00

5aPP1. Human middle-ear muscles rarely contract in anticipation of acoustic impulses. Heath Jones (APPD, USAARL, Bldg. 6901, Fort Rucker, AL 36362, heath.g.jones2.ctr@mail.mil), Nate Greene (APPD, USAARL, Aurora, CO), and William A. Ahroon (APPD, USAARL, Fort Rucker, AL)

The Auditory Hazard Assessment Algorithm for Humans (AHAAH) is an electrical equivalence model of the human ear designed to predict the risk for auditory injury from a given impulse noise exposure. One concern with the model is that the middle-ear muscle contraction (MEMC) associated with the acoustic reflex is implemented as a protective mechanism for certain instances in which a person is "warned" prior to the impulse. The current study tested the assumption that the MEMC can be elicited by a conditioning stimulus prior to sound exposure (i.e., a "warned" response). In order to quantify the MEMC, we used laser-Doppler vibrometry to measure tympanic membrane motion in response to a reflex-eliciting acoustic impulse. After verifying the MEMC, we attempted to classically condition the response by pairing the reflex-eliciting acoustic impulse (unconditioned stimulus, UCS) with various preceding stimuli (conditioned stimulus, CS). Changes in the magnitude and/or time-course of the MEMC following repeated UCS-CS pairings would be evidence of MEMC conditioning. Out of the 55 subjects tested so far, both non-conditioned ($n = 53$) and conditioned ($n = 2$) responses have been observed. These findings suggest that a "warned" MEMC may not be present in enough people to justify inclusion for being considered protective.

8:15

5aPP2. Neural mechanisms of selective listening to pitch and timbre in musical sounds. Yang Zhang, Zach Grice-Patil, Phillip Burton, and Cheryl Olman (Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, zhang470@umn.edu)

Previous research showed that different dimensions of sound such as pitch and timbre elicit distinct patterns of neural activities. Here, we report event related potential (ERP) along with functional magnetic resonance imaging (fMRI) data during a selective listening task of musical sounds. The participants were 12 normal adults. The stimuli were 4 musical tones with timbre (piano vs. violin) and pitch (high vs. low) contrasts. The stimuli were normalized for duration and intensity. In the timbre condition, the listener needed to indicate the instrument category. In the pitch condition, the listener needed to indicate the pitch level. While both conditions reached the ceiling-level accuracy, the pitch condition required significantly longer reaction time. Significant fMRI activations were found in the superior temporal gyri with right dominance for both conditions. The right superior temporal gyrus and left inferior frontal regions were important for coding pitch whereas the anterior cingulate cortex was exclusively active in selectively attending to timbre. These data demonstrate how different cortical networks contribute to selective listening. The ERP data further showed larger N1 and N2 peaks for the timbre condition than the pitch condition, suggesting that attentional modulation operates at both early and later stages of processing for selective listening.

8:30

5aPP3. Neurofeedback training of auditory selective attention enhances speech-in-noise understanding. Inyong Choi, Caroline Emory, Subong Kim (Commun. Sci. and Disord., Univ. of Iowa, 250 Hawkins Dr., Iowa City, IA 52242, inyong-choi@uiowa.edu), and Adam Schwalje (Otolaryngol. - Head & Neck Surgery, Univ. of Iowa Hospitals and Clinics, Iowa City, IA)

Speech-in-noise (SiN) understanding involves multiple cortical processes including feature extraction, grouping, and selective attention. Among those processes, we aimed to investigate the causal relationship between auditory selective attention and SiN performance. Selective attention enhances the strength of cortical neural responses to attended sounds while suppresses the neural responses to ignored sounds, which forms an evidence of sensory gain control theory. We hypothesized that the cortical response-guided neurofeedback could strengthen the sensory gain control, which in turn will improve selective attention performance and may result in better SiN understanding. With a single-blinded, between-subjects design including a placebo group, subjects were asked to attend to one of two simultaneous but asynchronous streams. For the participants assigned to the experimental group, a visual feedback was provided after each trial to demonstrate whether their attention was correctly decoded using their single-trial EEG response. The experimental group participants with four weeks of this neurofeedback training exhibited amplified cortical evoked responses to target speech as well as improved SiN understanding, while the placebo group participants did not show consistent improvement. To our best knowledge, this is the first report of selective-attention training enhancing SiN performance.

8:45

5aPP4. Effect of musical training on EEG measures of spectro-temporal processing. Coral Dirks (SLHS, Univ. of Minnesota, Minneapolis, MN), Anthony Shahin (Ctr. for Mind and Brain, Univ. of California, Davis, CA), Timothy Kwan (Biomedical Eng., Univ. of Malaya, Kuala Lumpur, Malaysia), and Evelyn E. Davies-Venn (SLHS, Univ. of Minnesota, 164 Pillsbury Dr SE, Minneapolis, MN 55445, venn@umn.edu)

Recent studies show that musical training enhances auditory processing abilities such as sensitivity to temporal fine structure and narrowband frequency resolution. Little is known about the effect of musical training on broadband spectro-temporal processing using rippled noise. This study evaluated whether musician enhancement in frequency resolution can be generalized to broadband spectro-temporal resolution using electrophysiological measures. We tested the hypothesis that musicians have enhanced broadband spectral resolution abilities compared to non-musicians. Spectral processing was measured using P1 N1 P2 responses to spectro-temporally modulated rippled noise. The test stimulus was spectro-temporally modulation wideband (4 octaves) rippled noise from 350 to 5600 Hz. EEG recordings were made using a 32-channel standard 10/20 configuration BioSemi system with a sampling rate of 4 kHz. For each condition, 1000 repetitions of the test stimuli were presented at 100% modulation with 2 Hz temporal modulation via electrically shielded ER-3A earphones, using a passive listening paradigm. The amplitude and latency of the P1-N1-P2 components

were measured to determine whether musicians had better sensitivity to spectro-temporal modulation compared to age-matched non-musicians. Results to date show slight musician enhancement in spectral processing. The clinical implications of these findings for auditory-rehabilitation will be discussed.

9:00

5aPP5. Timing effects in auditory-related forward predictions. Leonard Varghese and Barbara Shinn-Cunningham (Dept. of Biomedical Eng., Boston Univ., 610 Commonwealth Ave., Boston, MA 02215, lennyv@bu.edu)

Violation of learned sensory consequences may generate neural "error" signals that inform subsequent behavior. We investigated how this feedback loop might function in human audition, focusing specifically on timing mismatch signals in cortical structures. Participants pressed a button to trigger a sound at an expected, delayed onset. This sound was usually a broadband click ("standard"), but sometimes a low-pass filtered click ("target"). During 100% of trials during the first two experimental blocks, and on >80% of subsequent trials, these sounds occurred ~1 s after a button press. On the remaining <20% of trials, standards were presented slightly early ("early deviants") or slightly late ("late deviants") by 150–200 ms. Listeners were told to respond with a button press to targets; they were not informed about the timing manipulation. We used electroencephalography (EEG) to compare responses evoked by the physically identical expected standards, early deviants, and late deviants. EEG responses evoked by standards and temporal deviants appeared to reflect inter-subject differences in baseline auditory encoding strength, differences in mismatch encoding strength, and/or motor-induced suppression. Subsequent analyses will attempt to disambiguate the relative strength of these contributions, with the goal of determining the extent to which forward sensory predictions in audition reflect expected timing.

9:15

5aPP6. Midbrain sensitivity to frequency "chirps:" Implications for coding voiced speech sounds. Laurel H. Carney, Langchen Fan, and Kenneth S. Henry (Univ. of Rochester, 601 Elmwood Ave., Box 603, Rochester, NY 14642, Laurel.Carney@Rochester.edu)

Many neurons in the mammalian midbrain (inferior colliculus, IC) have strong selectivity for direction of frequency transitions ("chirps"), which often co-exist with the better-known sensitivity for amplitude-modulation (AM) frequency. Chirps occur when harmonics with a phase gradient are summed, as in Schroeder-phase harmonic stimuli. Chirps also occur in voiced speech due to the phase properties of vocal tract resonances. IC neurons are direction-selective for chirps in stimuli with fundamental frequencies in the voice-pitch range, and chirp selectivity can be achieved in an AM-tuned IC model by adding off-CF inhibition. Here, we will show physiological and model responses illustrating how chirps influence IC responses to speech in awake rabbit. In healthy auditory-nerve (AN) responses, the frequency extent, or size, of chirps vary with proximity of AN tuning to formant peaks due to inner ear nonlinearities. For example, synchrony capture causes one harmonic to dominate responses of AN fibers near spectral peaks, and responses that are dominated by one harmonic do not exhibit chirps. The systematic variation in chirp size across frequency channels and strong selectivity of IC neurons for chirps suggest a role for this feature in coding voiced speech. Importantly, this coding mechanism is vulnerable to sensorineural hearing loss.

9:30–9:45 Break

9:45

5aPP7. Improving acquisition time of the frequency-specific auditory brainstem response through simultaneous independently random toneburst sequences. Mark S. Orlando (Dept. of Otolaryngol., Univ. of Rochester, Rochester, NY) and Ross K. Maddox (Departments of Biomedical Eng. and Neurosci., Univ. of Rochester, 601 Elmwood Ave., Box 603, Rm. 5.7425, Rochester, NY 14642, ross.maddox@rochester.edu)

The frequency-specific auditory brainstem response (ABR) is an objective electroencephalographic (EEG) method for estimating auditory thresholds when behavioral thresholds are not attainable, primarily in infants and toddlers. It is recorded by repeatedly presenting brief, band-limited tonebursts and averaging the scalp potential response. In a typical ABR evaluation, measurements are made at multiple intensities, at multiple frequencies, and in both ears, leading to a large combinatorial space and long test times. Here, we present a method of recording the responses at five frequencies in both ears simultaneously, along with preliminary data demonstrating its effectiveness. We created ten toneburst trains—five octave frequencies (500–8000 Hz) by two ears—for simultaneous presentation. Importantly, the toneburst timing for each train was dictated by an independent random process, rather than being periodic. Independent sequences allow separate computation of the ABR to each of the toneburst trains as the cross-correlation of its timing sequence and the EEG recording. Early results suggest substantial improvements in recording time, particularly at lower stimulus intensities. It also appears that cochlear spread of excitation—a problem at high intensities—is mitigated by our method. This improvement in place specificity is predicted by modeling of auditory nerve activity.

10:00

5aPP8. On the relationships between auditory evoked potentials and psychoacoustical performance. Dennis McFadden, Edward G. Pasanen, Mindy M. Maloney (Psych., Univ. of Texas, Austin, 108 E. Dean Keeton A8000, Austin, TX 78712-1043, mcfadden@psy.utexas.edu), Erin M. Leshikar, Michelle H. Pho, and Craig A. Champlin (Dept of Commun. Sci. & Disord., Univ. of Texas, Austin, TX)

Performance was measured on several common psychoacoustical tasks for about 70 subjects. The measures included simultaneous and temporal masking, masking by tones and by complex sounds, critical bandwidth, release from masking, and detection in the quiet. Also measured were auditory evoked potentials (AEPs, short and middle latency). Of interest were the correlations between psychoacoustical performance and the AEP measures, as well as any mean differences in the AEPs by sex and by menstrual cycle. Subjects were tested behaviorally in same-sex crews of 4—8 members; behavioral testing required from 8–10 weeks for each crew. Correlation and effect size were the primary measures of interest. Resampling was used to determine implied significance for the various comparisons studied. (This lab previously reported race differences in psychoacoustical performance and in otoacoustic emissions for these same subjects.) Several AEP measures exhibited large sex differences that were similar across race. Some correlations between psychoacoustical tasks and the various AEP measures were moderately high, but, across sex and race, the individual differences observed in psychoacoustical performance generally were not compellingly related to the individual differences in AEP latency or amplitude. N.F. Viemeister never worked on any of these topics. [Work supported by NIH/NIDCD.]

10:15

5aPP9. Behavioral and electrophysiological evidence of incidental learning, generalization, and retention of speech categories from continuous speech. Yunan C. Wu (Psych., Carnegie Mellon Univ., Psychology-Baker Hall, 5000 Forbes Ave., Pittsburgh, PA 15213, charleswu.01@gmail.com), Ran Liu (MARi, Pittsburgh, PA), Sung-Joo Lim (Dept. of Speech and Hearing Sci., Biomedical Eng., Boston Univ., Boston, MA), and Lori L. Holt (Psych., Carnegie Mellon Univ., Pittsburgh, PA)

Speech learning involves discovering appropriate functional speech units (e.g., speech categories) embedded in a continuous stream of speech. However, speech category learning has been mostly investigated with isolated sound tokens. Here, we used a videogame to encourage incidental learning of speech categories from continuous speech input (Lim *et al.*, 2015). Native English participants (N = 17) played the videogame while listening to acoustically-variable continuous Mandarin sentences. Unbeknownst to participants, four acoustically-variable Mandarin keywords were embedded in the continuous sentences. During training, participants were not informed about the keywords, made no overt categorization decisions, and received no feedback. Participants' post-training categorization test demonstrated robust incidental learning of keyword that persisted even 10 days after training, and generalized to novel utterances and talkers. Further, the N100 response in the frontal EEG site evoked by keyword onsets within continuous Mandarin speech during passive listening was greater post-training compared to pre-training. This neural enhancement was specific to the Mandarin keywords functionally useful in videogame training. Our results demonstrate that although participants were not informed about the keywords, they did not make overt categorization decisions during videogame play, they incidentally learn functionally-relevant non-native speech categories from continuous speech input across considerable acoustic variability.

10:30

5aPP10. Sequential stream segregation based on spatial cues: Behavioral and neural measurements. Marion David and Andrew J. Oxenham (Psych., Univ. of Minnesota, 75 East River Parkway, Minneapolis, MN 55455, david602@umn.edu)

Differences in simulated spatial cues in the horizontal plane have been shown to enhance both voluntary and obligatory stream segregation of sounds with realistic spectro-temporal variations, such as sequences of syllables. In this experiment, listeners were presented with sequences of speech tokens, each consisting of a fricative consonant and a voiced vowel. The CV tokens were concatenated into interleaved sequences that alternated in simulated spatial positions. The interleaved sequences lasted 1 min. The listeners had to press a button each time they heard a repeated token. In the selective attention task, the listeners were asked to attend only one of the two interleaved sequences; in the global attention task, the listeners had to perceive the interleaved sequences as single stream to detect a repetition between the sequences. Simultaneous EEG measurements were made. The behavioral results confirmed that listeners were able to either attend selectively or globally, depending on the task requirements. The EEG waveforms differed between the two tasks, despite identical physical stimuli, reflecting the difference between global and selective attention. Both behavioral and EEG results reflected the effects of increasing spatial separation in enhancing selective attention and making global attention to the sequences more difficult.

10:45

5aPP11. Behavioral measures of cochlear gain reduction and gain reduction in with normal hearing or minimal cochlear hearing loss. Elizabeth A. Strickland, Hayley Morris, Miranda Skaggs, William Salloom, 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 various psychoacoustic measures thought to be related to cochlear function. In the listeners tested, 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 (MOCR) decreases the gain of the cochlear active process in response to sound, when possible, measures were made with short stimuli. Signal frequencies were 2, 4, and 8 kHz. Maximum gain was estimated by measuring the threshold masker level for a masker at the signal frequency and a masker well below the signal frequency. The effect of signal duration on threshold was measured using 10 and 200-signals. One point on the lower leg of the input/output function was measured by finding the threshold masker level for a masker slightly less than one octave below the signal frequency needed to mask a signal at 5 dB SL. Gain reduction was estimated by presenting a pink noise precursor before the signal and masker, and measuring the change in signal threshold as a function of precursor level. The relationship between these measures will be discussed. [Work supported by NIH(-NIDCD)R01 DC008327 (EAS), and grants from the Purdue Office of the Executive Vice President for Research and the Purdue Graduate School (WS).]

Session 5aSC

Speech Communication: Speech Production (Poster Session)

Marc Garellek, Chair
UCSD, 9500 Gilman Drive #0108, La Jolla, CA 92093-0108

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 authors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

5aSC1. Speech formant changes due to repeated measurements, instruction, and simulated environments using both automated and manual feature extraction. Mark L. Berardi (Dept. of Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., Rm. 211D, Oyer Speech & Hearing Bldg., East Lansing, MI 48824, berardi1@msu.edu), Jessica J. Staples, Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT), and Eric J. Hunter (Dept. of Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI)

Speech production can differ depending on how speech is elicited (e.g., spontaneous speech, read text, speaking style instructions, the speaking environment). Previous studies have shown different speaking styles being elicited via instruction (e.g., clear speech) or via the speaking environment (e.g., Lombard speech). There is evidence that the acoustic features of clear speech elicited by reading are similar to those observed in semi-spontaneous interaction between two interlocutors, but that clear speech changes are of a greater magnitude in the read speech than the semi-spontaneous speech. Ten talkers (five male, five female) performed read sentences (BVD) and picture descriptions in several conditions. The present study compares vowel formants from BVD sentences and spontaneously produced picture descriptions in a variety of conditions. Conditions were as follows: (1) repeated measures given the same instructions, (2) the effects of two speaking style instructions (conversational and clear), and (3) four simulated listening environments (quiet, 55 dB SPL of white noise, 63 dB SPL of white noise, and a reverberant environment) presented via earphones. Formants were extracted both manually (hand marked) and using automated techniques. Acoustic features relevant to the speaking styles and simulated conditions will be discussed in terms of the two extraction techniques.

5aSC2. Source-filter interaction: A study using vocal-tract data of a soprano singer. Tokihiko Kaburagi, Momoyo Ando, and Yasufumi Uezu (Kyushu Univ., Shiobaru 4-9-1, Minami-ku, Fukuoka 815-8540, Japan, kabu@design.kyushu-u.ac.jp)

When the fundamental frequency of voice is close to a formant frequency, the voice-source system in the larynx and the acoustic filter of the vocal tract can be coupled acoustically. This source-filter interaction (SFI) can cause unsteadiness in vocal fold oscillations, sudden jump of the fundamental frequency, and transitions in voice register. In this study, a magnetic resonance imaging device was used to scan the vocal tract of a soprano singer in three dimensions when she produced sustained vowels with different notes ranging from A3 (220 Hz) to A5 (880 Hz). The cross-sectional area and the input impedance of the vocal tract were then determined from image data. As a result, the frequency and the magnitude of the first impedance peak for the /i/ vowel increased when the note became higher, suggesting that a strong source-filter interaction may take place for high notes when the fundamental frequency is close to the peak frequency. We therefore conducted a computer simulation of voice generation using models of the vocal folds and the vocal tract. Our preliminary results showed different SFI effects depending on the magnitude of the impedance peak. [This work was supported by JSPS KAKENHI Grant No. JP16K00242.]

5aSC3. Impact of phonatory frequency and intensity on glottal area waveform measurements derived from high-speed videoendoscopy. Rita R. Patel (Dept. of Speech and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405-7002, patelrir@indiana.edu), Michael Döllinger, and Stefan Kniesburges (University Hospital Erlangen, Med. School, Div. of Phoniatrics and Pediatric Audiol., Univ. Erlangen-Nürnberg, Erlangen, Bavaraia, Germany)

Measurements of glottal area waveform from high-speed videoendoscopy were made on vocally healthy females (n = 41) and males (n = 25) during sustained /i/ production at typical pitch and loudness, high pitch, and soft phonation. Three trials of each condition were performed yielding 594 samples. Statistical analysis of glottal cycle quotients (open quotient (OQ), speed quotient (SQ), rate quotient (RQ), glottal gap index (GGI)), glottal cycle periodicity (amplitude, time (TP)), glottal cycle symmetry (phase asymmetry index, spatial symmetry index, and amplitude symmetry index), glottal area derivative (maximum area declination rate (MADR)), and mechanical stress measures (stiffness index (SI), amplitude-to-length ratio (ALR)) revealed that only SI varied systematically across pitch and loudness conditions for males and females. Variations in pitch and loudness results in changes in SI, ALR, RQ, MADR, and SQ for females, whereas variations in target pitch and loudness results in changes in SI, ALR, RQ, MADR, GGI, OQ, and TP for male speakers. Carefully selecting the laryngeal parameters derived from high-speed videoendoscopy has the potential to provide insights clinically into the known laryngeal biomechanics expected for the increase in pitch and reduction in vocal loudness.

5aSC4. Voice instabilities in a three-dimensional continuum model of phonation. Zhaoyan Zhang (UCLA School of Medicine, 1000 Veteran Ave, 31-24 Rehab Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu)

The goal of this study is to identify vocal fold conditions that produce irregular vocal fold vibration and the underlying physical mechanisms. Using a three-dimensional computational model of phonation, parametric simulations are performed with co-variations in vocal fold geometry, stiffness, and vocal tract shape. For each simulation, the cycle-to-cycle variations in the amplitude and period of the glottal area function are calculated, based on which the voice are classified into three types corresponding to regular, quasi-steady or subharmonic, and chaotic phonation. The results show that the presence of a vocal tract significantly increases the occurrence of irregular vocal fold vibration, which naturally occurs even under symmetric vocal fold conditions, in particular for vocal folds with very low transverse stiffness in the coronal plane or a soft vocal fold body layer. The occurrence of voice instability is suppressed by increasing the transverse stiffness, increasing the body-cover ratio in the longitudinal stiffness, or decreasing the subglottal pressure. The implications of these observations on the production of certain voice qualities are discussed. [Work supported by NIH.]

5aSC5. Direct measurement of glottal flow waveform in an excised K9 larynx with a vocal tract. Alexandra Maddox, Liran Oren, Ephraim Gutmark, Charles P. Farbos de Luzan, and Sid M. Khosla (Univ. of Cincinnati, 3317 Bishop St., Apt. 312, Cincinnati, OH 45219, maddoxat@mail.uc.edu)

False vocal folds (FVF) or ventricular folds have been shown to impact the vibration of the true folds. Using computational and experimental models to examine the aerodynamic and acoustic effects the FVF, previous studies have shown that the presence of FVF lowers the phonation threshold pressure, the overall intraglottal pressure distribution, and enhances intraglottal vortical structures. However, the full effect of the FVF on the glottal flow waveform has never been measured in a tissue model of the larynx. Therefore, the objective of this study was to evaluate the impact of FVF on the glottal flow waveform in an excised canine model. A vocal tract model was placed over the larynx, and direct velocity measurements were taken at the glottal exit using tomographic particle image velocimetry. Measurements were taken with a systematic change of the FVF constriction and integrated to calculate the air flow waveform. The results show that a restriction above the folds increased the overall flow rate (dQ/dt) and change the maximum flow declination rate (MFDR). The 3-D measurements at the glottal exit can also be used to validate previous computational studies on how the vocal tract affects the phonation mechanism. The clinical significance of these findings will be further discussed.

5aSC6. Tongue muscle shortening patterns during speech. Maureen Stone, Natalie Leem, Katie N. Garret (Univ. of Maryland Dental School, 650 W. Baltimore St. Rm. 8207, Baltimore, MD 21201, mstone@umaryland.edu), Jonghye Woo (Massachusetts General Hospital, Boston, MA), and Jerry L. Prince (Johns Hopkins Univ., Baltimore, MD)

This study will used tagged-MRI to examine the 3D shortening patterns of the genioglossus, verticalis, and transversus muscles during speech tasks that move the tongue backwards, forwards, and in complex deformations. Tissue points at each muscle's origin, insertion, and points in between are used to calculate length changes over time. To enhance the range of subject differences, 10 controls and 5 glossectomy subjects will be examined. The motivation is to understand how muscle behavior is linked to internal tongue motion patterns. Activated muscles typically shorten along their line of action. In the tongue, however, muscles interdigitate, so that when one muscle shortens, others are passively pulled out of linear alignment. Thus, muscle fibers may not form a straight line when activated or at rest. In addition, the mechanical consequences of activating a specific muscle can be complex since the tongue moves continuously during speech and different regions of the tongue experience different demands. Moreover, since the tongue stretches in three-dimensions, some muscles may activate to augment a primary muscle, that is, to prevent expansion in the wrong direction. Muscle shortening from tagged MRI does not capture actual muscle activity, but can provide insight into interactive muscle behaviors.

5aSC7. Automatic tongue contour extraction in ultrasound images with convolutional neural networks. Jian Zhu, Will Styler (Dept. of Linguist, Univ. of Michigan, 1052 Island Dr., Apt 104, Ann Arbor, MI 48105, zhujianbw@gmail.com), and Ian C. Calloway (Dept. of Linguist, Univ. of Michigan, Ypsilanti, MI)

Ultrasound imaging of the tongue can provide detailed articulatory information addressing a variety of phonetic questions. However, using this method often requires the time-consuming process of manually labeling tongue contours in noisy images. This study presents a method for the automatic identification and extraction of tongue contours using convolutional neural networks, a machine learning algorithm that has been shown to be highly successful in many biomedical image segmentation tasks. We have adopted the U-net architecture (Ronneberger, Fischer, & Brox 2015, U-Net: Convolutional Networks for Biomedical Image Segmentation, DOI:10.1007/978-3-319-24574-4_{28}), which learns from human-annotated splines using multiple, repeated convolution and max-pooling layers in the network for feature extraction, as well as deconvolution layers for generating spatially precise predictions of the tongue contours. Trained using a preliminary dataset of 8881 human-labeled tongue images from three speakers, our model generates discrete tongue splines comparable to those identified by human annotators (Dice Similarity Coefficient = 0.71). Although work is ongoing, this neural network based method shows considerable promise for the post-processing of ultrasound images in phonetic research. [Work supported by NSF grant BCS-1348150 to P.S. Beddor and A.W. Coetzee.]

5aSC8. A method for distinguishing tongue surface topology for different categories of speech sound. Jonathan N. Washington (Linguist. Dept., Swarthmore College, 500 College Ave., Swarthmore, PA 19081, jwashin1@swarthmore.edu) and Paul A. Washington (Dept. of Petroleum Eng. and Geology, Marietta College, Marietta, OH)

An analytic method has been developed to assist in interpreting data from two-dimensional ultrasound tongue imaging by determining whether the position of the tongue surface significantly differs for two categories of speech sound and, if so, which region is most associated with the difference. Individual tongue traces are modeled as curves in a polar coordinate system around a reference point. Two categories of speech sound are compared as a function of the distance between the traces of the two categories at each arc angle. The regions of the arc with the most significant deviation (i.e., where the categories are best contrasted) are shown by the largest ratio of mean to standard deviation (the greatest number of standard deviations separating the mean tongue surfaces). Categories are most distinguished in regions with more than one standard deviation separating them. A ratio of the angular distance between a point of complete overlap (no contrast) and the maximum positive and negative differentiation points has potential as a speaker-agnostic measure of which regions of tongue position are important at the level of the linguistic variety. To demonstrate this method, its application to the vowel anteriority contrast in several Turkic languages is presented.

5aSC9. Palate shape and articulatory range. Keith Johnson (Linguist, UC Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94720, keithjohnson@berkeley.edu)

Using the Wisconsin X-ray microbeam corpus, this paper explores the relationship between palate shape in the saggital plane and articulatory variation. The corpus has been updated with tags (Praat Textgrids) marking the acoustic onsets and offsets of words and phones. In addition, the sound files have been translated to WAV format, and the articulatory data saved as plain text files. The shape of the palate for each of 57 speakers of American English was measured in terms of the total 2D area of the palate vault above the occlusal plane. Articulatory variability was measured in terms of the interquartile ranges of the x and y locations of 7 pellets attached to the tongue (4 pellets), lips (upper and lower), and jaw. Although there is a significant correlation between palate area and the range of motion of some articulators (particularly on the body of the tongue, but surprisingly to this author, not the motion of the jaw), the study also finds that there is a great deal of variation among speakers. Articulatory motions involved in particular speech sounds will also be examined to determine if there are specific patterns of variation associated with the palate size.

5aSC10. Acoustic correlates of prominence in Yawarana. Natalia Cáceres Arandia, Alyssa Moore, Zac Post, Spike Gildea, and Melissa M. Baese-Berk (Dept. of Linguist, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, mbaesebe@uoregon.edu)

Yawarana (Cariban) is a critically endangered language of Venezuela, with 20–30 speakers and little published research beyond wordlists. In these wordlists, orthographic indications of stress and/or vowel length are inconsistent. As part of our language documentation project, we examine acoustic correlates of prominence in Yawarana. Many other languages in the Cariban family have a clear rhythmic stress system, with vowel lengthening and pitch excursion marking prominence; Yawarana does not mark prominence in the same way. Even so, in some situations, native speakers do appear to attend to stress (i.e., when correcting the pronunciation of language learners). Here, we ask what the acoustic correlates of prominence are, paying particular attention to intensity, pitch, and duration information. We are examining over 200 lexical items produced by four native speakers. These items are repeated in isolation, produced in carrier phrases, and produced in narratives and conversations. We compare measures of prominence in each of these situations. This investigation will pave the way for a more in-depth study of acoustic features of Yawarana and will inform future studies of prominence in other Cariban languages.

5aSC11. Acoustic measures of delabialized velars in Hong Kong Cantonese. 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)

Hong Kong Cantonese is undergoing a neutralization of the labialized velar /k^{w}/ with the plain velar /k/ before the round back vowel /ɔ/ that is well-documented auditorily but not acoustically (Bauer, 1982; Newman, 1987; To *et al.*, 2015). The present data from 14 native speakers of Hong Kong Cantonese seeks to determine what acoustic characteristics differentiate non-neutralized /k/ and /k^{w}/, and whether neutralization of these segments in neutralization environment is complete or incomplete. Initial data analysis suggests that locus equations on the second and third vowel formants may not be reliable cues in distinguishing plain and clearly labialized (non-merged) velars in this dialect of Cantonese, although formant lowering and increased vowel length for non-merged, labialized productions appear consistent, and intensity measures at 50 ms post-vowel onset may prove to be an important cue as well. The most consistent cues for labialization in clearly labialized articulations of target words will then be measured in neutralized (phonetically plain) productions in order to determine whether any acoustic traces of labialization remain for these realizations.

5aSC12. The position of the tongue root in the articulation of posterior sibilants in Polish. Malgorzata E. Cavar (Dept. of Linguist, Indiana Univ., Ballantine Hall 844, 1020 E. Kirkwood Ave., Bloomington, IN 47405-7005, mcavar@indiana.edu) and Steven M. Lulich (Speech and Hearing, Indiana Univ., Bloomington, IN)

Standard Polish has a typologically relatively rare contrast between "hard" post-alveolar affricates and fricatives, transcribed by Ladefoged and Disner (2012) as (/tṣ, dẓ, ṣ, ẓ/ and "soft" alveolo-palatal affricates and fricatives (IPA transcription /tɕ, dʑ, ɕ, ʑ /). The hard post-alveolars are notoriously ambiguous—phonetically they are neither sense stricto retroflexes (though some authors adopt such an analysis based on phonological arguments, e.g., Hamann 2003) nor typical palatoalveolars. Additionally, the "hard" post-alveoalars can be allophonically palatalized in the context of [i]. Multiple approaches have been proposed to differentiate the hard and soft series, all of them primarily focusing on the shape of the body of the tongue, including the level of raising of the body of the tongue, the place of articulation, and the length of the constriction. In this study, we present 3-D tongue shapes of the consonants. The 3D ultrasound images—besides additional details of the shape of the tongue body—reveal a consistent difference in terms of the tongue root position, with a fronting of the tongue root and a pronounced groove along the center of the tongue in the root part of the tongue for prepalatals and no such fronting for the "hard" post-alveolars.

5aSC13. Creaky phonation and Mandarin 214-Sandhi. Noah Elkins (Linguist, Macalester College, 1731 Dayton Ave., Apartment 4, St. Paul, MN 55105, nelkins@macalester.edu)

Mandarin has four contrastive tones (55, 35, 214, and 51). The 214 tone is consistently produced at the lowest part of a speaker's pitch range, and is therefore frequently accompanied by allophonic creaky phonation. Additionally, Mandarin has a complex system of tone sandhi; the type of sandhi being investigated in this paper is whereby, when two adjacent 214 tones are uttered, the first 214 changes to a 35, and the second 214 changes to a 21. Twenty speakers of Mandarin were recorded at the campus of Peking University using a sentence list constructed to elicit minimal pairs between sandhi-unaffected and sandhi-affected 214 tones. Age and gender were not considered a contributing factor to the presence of 214-linked creak, and so such factors were not controlled for. Preliminary results show that there is no significant difference between the amount of creak on affected and unaffected vowels. Therefore, the duration spent uttering the vowel at a lower pitch did not increase the intensity of creak. However, the continued presence of creak on a sandhi-affected 214 vowel may serve as a perceptual cue to that tone, whereby a Mandarin speaker can understand the meaning of the syllable even without a robust pitch contour.

5aSC14. The effects of prosody on pitch and voice quality of White Hmong tones. Marc Garellek (Linguist, UCSD, 9500 Gilman Dr. #0108, La Jolla, CA 92093-0108, mgarellek@ucsd.edu) and Christina M. Esposito (Linguist, Macalester College, St. Paul, MN)

White Hmong contrasts two high-falling tones (one breathy, the other modal) and two low tones (one modal level-tone, the other creaky low-falling). Perceptual studies [Garellek *et al.* (2013)] have shown that listeners rely on breathy voice to distinguish between the high-falling tones, but ignore creaky voice when distinguishing between the low tones. We test whether such differences stem from prosodic variation, by examining tokens from stories read by native speakers. Vowels were annotated for phrasal position and neighbouring tones. We obtained f0 and voice quality measures. Results support and help elucidate previous perceptual research: (1) the breathy high-falling tone is breathy in all prosodic positions, (2) the breathy high-falling tone has a prosodically-variable f0, (3) the creaky low-falling tone has a prosodically-stable f0. The creaky low-falling tone is creakier than the modal low tone in all positions. Therefore, listeners may ignore f0 in the identification of the breathy high-falling tone because its pitch is prosodically variable. However, the creaky low-falling tone is consistently low-falling, which could explain why listeners rely on f0 as the dominant cue. Discussion will include why listeners ignore creaky voice if it is robust in production.

5aSC15. Acoustic phonetic measurements of grammatical tones in Anyi. Ettien N. Koffi (English, Saint Cloud State Univ., 720 Fourth Ave. South, Saint Cloud, MN 56301, enkoffi@stcloudstate.edu)

Talkers and hearers of Anyi, a West African language of the Akan family, are very adept at discriminating between sentences containing verbs conjugated in the indicative, the intentional, and the subjunctive moods just by relying on subtle variations in pitch. A phonetic investigation is undertaken to determine as precisely as possible the acoustic cues that contribute the most to intelligibility. Ten speakers produced five sentences each in all three moods for a total of 150 utterances. The tone bearing units (TBU) on the subject pronoun and the disyllabic verbs are analyzed acoustically. The acoustic correlates investigated are F0, intensity, and duration. All in all, 810 TBUS are measured. The main findings are as follows. The F0 of the subject pronoun and of the verb are the most robust cues for discriminating between the declarative and the intentional moods. The differentiation between the intentional and the subjunctive moods rests principally on the F0 of the subject pronoun. The distinction between the indicative and the subjunctive moods depends mainly on the F0 of the verb and to a lesser extent on duration. Intensity does not seem to be a robust cue for discriminating between these three tone-induced grammatical constructions.

5aSC16. Timing patterns of voiceless and voiced singleton and geminate plosives of Yanagawa Japanese. Shigeko Shinohara (Lab. of Phonet. and Phonology, CNRS/Univ. of Sorbonne-Nouvelle, Paris, France), Qandeel Hussain (Dept. of English (Linguist Program), North Carolina State Univ., 221 Tompkins Hall, Campus Box 8105, Raleigh, NC 27695-0001, qhussai@ncsu.edu), and Masako Fujimoto (Adv. Res. Ctr. for Human Sci., Waseda Univ., Tokorozawa, Saitama, Japan)

The paper examines acoustic timing patterns of word-medial voiceless and voiced singleton and geminate plosives of Yanagawa Japanese, one of the Chikugo varieties of Japanese spoken in the center of Kyushu (Japan). Unlike standard Tokyo Japanese, Yanagawa Japanese is characterized by frequent gemination of any types of consonants including voiced obstruents (e.g., /kuzzoko/ [kuddzoko] "sole, the fish," /mirogge/ [miɾogge] "a shell fish"). Five Yanagawa Japanese speakers were recorded. Another five speakers of standard Tokyo Japanese were also recorded as a control group. The stimuli consisted of nonsense words with $C_1V_1C(C)_2V_2$ structure (e.g., /kak(k)a/, /kag(g)a/). The findings suggest that the whole word duration containing geminate consonants was longer in proportion to mora count difference those containing singleton consonants, confirming moraic timing at word level in Yanagawa Japanese. When C_2 was a geminate (voiceless or voiced), it lengthened the preceding vowel (V_1) and shortened the following one (V_2) in both dialects. However, the magnitude of influence of geminates to adjacent vowels was dialect specific: V_1 duration in Yanagawa was consistently longer than Tokyo when the following consonant was a geminate.

The results of the current study point towards the timing differences in singletons and geminates across Japanese dialects.

5aSC17. Phonetic realization of vowel reduction in Brazilian Portuguese. Sejin Oh (Linguist, The Graduate Ctr., CUNY, 365 5th Ave., New York, NY 10016, soh@gradcenter.cuny.edu)

This study explored the phonetic realization of vowel reduction in Brazilian Portuguese, which reportedly exhibits raising of non-high vowels in unstressed syllables. Specifically, we tested the influence of speech rate on the realization of /a/ in five prosodic positions: word-initial pretonic, medial pretonic, tonic, medial posttonic, and final posttonic. The results showed that, while both speech rate and prosodic position had a clear effect on the phonetic duration of vowels, F1 values were far better predicted by the vowel's prosodic position (non-posttonic vs. posttonic), although some effects of speech on F1 were observed in vowels at word edges. Correlations between phonetic duration and F1 were statistically significant but generally weak in all positions. We argue these findings suggest that vowel reduction in Brazilian Portuguese primarily reflects phonological patterning rather than phonetic undershoot, although phonetic reduction is also apparent. We discuss the results in the context of cross-linguistic studies of vowel reduction, and the relation between phonetics and phonology.

5aSC18. Phonetic variability of nasals and voiced stops in Japanese. Yoichi Mukai and Benjamin V. Tucker (Dept. of Linguist, Univ. of AB, 3-26 Assiniboia Hall, Edmonton, AB T6G2E7, Canada, mukai@ualberta.ca)

We investigated the phonetic variability of nasals and voiced stops in a large-scale Japanese speech corpus. We then analyzed the types of variation across speech styles. In particular, we examined the instances where target segments are deleted or realized as different phonemes. We identified 285 lexical entries displaying variability of target segments in the Corpus of Spontaneous Japanese (Maekawa, 2003). In line with the findings of Arai (1999), we observed a few variants of voiced stops, such as /d/ becoming [n] in /doɴna/ "what" and /g/ being deleted in /daigaku/ "university." We also found /b/ turning into [m], such as /boku wa/ "I am" becoming [mo kaː]. For nasals, we observed /zeɴiɴ/ "all the people" and /geɴiɴ/ "reason" becoming [zẽːɪɴ] and [gẽːɪɴ], in which the first /ɴ/ was deleted and the preceding /e/ was nasalized and lengthened. Our findings suggest that the extent to which speakers produce phonetic variants of target segments is likely specified lexically more than stylistically because we find instances of [zẽːɪɴ] and [gẽːɪɴ] produced across multiple speech styles. Further, we find more than 90% of the instances of /geɴiɴ/ were pronounced as [gẽːɪɴ] and 80% of /zeɴiɴ/ were pronounced as [zẽːɪɴ].

5aSC19. Stress-sensitive consonant gemination through plural noun reduplication in Tohono O'odham. Daejin Kim and Robert Cruz (Linguist, Univ. of New Mexico, 1 University of New Mexico, MSC03 2130, Albuquerque, NM 87131-0001, daejinkim@unm.edu)

This study examines how plural noun reduplication in Tohono O'odham (TO) language is phonetically realized. To make a noun plural, the first CV sequence (i.e., base; e.g., *goks* "a dog" → *go**go**ks* "dogs") or the first consonant of the base (e.g., *pado* "a pig" → *pa**p**do* "pigs") are reduplicated after the base (Hill & Zepeda, 1998). Base and reduplicant have been regarded phonologically equivalent (i.e., consonant gemination; Fitzgerald, 2003). In a sense that TO has the strong-weak stress pattern across syllables (e.g., *tówa* "turkey"), stress may influence reduplicants' duration. The CV reduplicant at the second syllable with the weak stress may not be phonologically same as the base and the C reduplicant at the coda of the base with the strong stress. If CV and C reduplicants are equivalent even with stress, an additional prosodic unit (i.e., mora) may exist. The analysis of TO speech supports that the base (CV_1) is longer than the reduplicant (CV_2 or C_2) even with other contextual variables. Reduplication process may be stress-timed, but it is unknown whether speakers equally treat base and reduplicants. This paper will also discuss a need to examine how TO speakers control the timing with rhythm between base and reduplicant.

5aSC20. Place of articulation effects on voice onset time and phonation bleed persist in languages with no voicing distinction. Stephanie Kakadelis (Linguist, The Graduate Ctr., CUNY, 365 Fifth Ave., New York, NY 10016, skakadelis@gradcenter.cuny.edu) and D. H. Whalen (Linguist, The Graduate Ctr., CUNY, New Haven, CT)

Place of articulation effects on voice onset time (VOT) and phonation bleed in the closure of oral stops has been observed in languages which utilize these as perceptual correlates to voicing distinctions (Lisker & Abramson, 1967; Ohala & Riordan, 1979). In this study, measurements from intervocalic oral stops were collected from three No Voicing Distinction (NVD) languages, Bardi (ISO-639 bcj), Arapaho (ISO-639 arp), and North Puebla Nahuatl (ISO-639 ncj). Tokens surfacing with positive VOT increased in duration as place of articulation went from more anterior to dorsal. VOT for Arapaho coronal oral stops was longer on average than velar orals stops. Nahuatl showed a similar pattern, with average VOT increasing from labial to coronal to velar. Average VOT for Arapaho labial tokens and all Bardi tokens was negative, so they were not included in VOT measurements. Conversely, average duration of phonation bleed in all three languages was greatest for labial oral stops, decreasing in average duration for coronals and velars. This suggests a physiological effect of vocal tract volume on both VOT and maintenance of phonation during the closure of an oral stop even in the absence of a phonological distinction.

5aSC21. Inter- and intra-speaker variability in the production of voiceless nasal consonants. Maureen Hoffmann (Dept of Linguist, Univ. of Arizona, P.O. Box 210025, Tucson, AZ 85721, mhoffm@email.arizona.edu)

This study uses both acoustic and aerodynamic data to investigate variation in the realization of phonemically voiceless nasal consonants, which have been found to differ cross-linguistically. In particular, it examines the timing of voicing and oral and nasal air flow during production of voiceless nasal consonants in both word-initial and word-medial positions in Hakha Chin, a Tibeto-Burman language spoken in western Burma/Myanmar. Comparing three other Tibeto-Burman languages, Bhaskararao and Ladefoged (1991) found two distinct patterns in the production of voiceless nasal consonants, which varied by language, but reported general consistency among speakers of the same language. Results of the present study show considerable variation in the realization of voiceless nasals, both between individual speakers and even between individual tokens produced by the same speaker. This high degree of variability may be linked to the relatively weak acoustic cues for place of articulation among nasal consonants in general and for voiceless nasals in particular.

5aSC22. The effect of lexical competition on vowel duration before voiced and voiceless English stops. Eleanor Glewwe (Univ. of California, Los Angeles, 3125 Campbell Hall, Los Angeles, CA 90095-1543, eleanorglewwe@gmail.com)

One source of phonetic variation is lexical competition, which has been shown to cause both hyperarticulation, especially of vowel formants (e.g., Wright 2004), and reduction, manifested as shorter segment and word durations (Kilanski 2009, Gahl *et al*. 2012). Studies have found that lexical competition (e.g., having a minimal pair competitor, having more phonological neighbors) causes contrastive hyperarticulation of the initial stop voicing contrast in English: greater competition makes VOT longer for voiceless stops and/or shorter for voiced stops (Baese-Berk & Goldrick 2009, Nelson & Wedel 2017). I conducted a corpus study that looked for contrastive hyperarticulation of the final stop voicing contrast in English. The cue examined was preceding vowel duration. I did not find contrastive hyperarticulation: neither minimal pair competitor existence nor higher neighborhood density caused vowels to be longer before final [d] or shorter before final [t]. Instead, both competition metrics correlated with shorter vowel durations before [t] and [d]. This result is consistent with Goldrick *et al*. (2013)'s failure to find contrastive hyperarticulation of the final stop voicing contrast and their hypothesis that competition affects initial and final contrasts differently. It is also consistent with Gahl *et al*'s (2012) finding that high neighborhood density causes reduction.

5aSC23. The effect of domain-initial strengthening on voice onset time and vowel length on English oronyms. Marisha D. Evans (Linguist, Macalester College, 948 North St., Ste. 5, Boulder, CO 80304, mevans3@macalester.edu)

Oronyms are sets of words and phrases that contain the same phonemes but differ in word boundary (*atop*, *a top*). Past research established that domain-initial strengthening (DIS) affects segments at the beginning of prosodic domains. This research examines voice onset time and vowel length in English oronyms with the hope of finding a systematic strengthening of the target sounds word-initially. The word list consists of oronym pairs with the same phonemes and the same [vowel, voiceless stop, vowel] at the beginning of the oronym (e.g. *atop*/*a top*, *attack*/*a tack*). To elicit more natural-sounding tokens, participants created sentences with the target oronym at the beginning, or read already-prepared sentences. It is hypothesized that there will be a systematic strengthening of the initial segments (in the form of either a longer VOT or vowel length). This could mean speakers are fully aware of word boundaries and use DIS to help avoid lexical ambiguity when speaking.

5aSC24. The effect of cognitive load on the vowel onsets of older and younger adults. Richard J. Morris, Megan K. MacPherson, and Maria Roa (Commun. Sci. and Disord., Florida State Univ., 201 West Bloxham Rd., 612 Warren Bldg., Tallahassee, FL 32306-1200, richard.morris@cci.fsu.edu)

The purpose of the study was to determine the effect of cognitive load on the vowel timing of older and younger adults. It was hypothesized that the participants would have longer and less extensive second formant (F2) transitions in the heavier cognitive load condition. It was also hypothesized that the older adults would have longer and less extensive F2 transitions than the younger adults. Eight adults, four younger and four older, equally matched for sex participated. They completed a sentence-level Stroop task in two conditions, congruent and incongruent, that varied on cognitive load. Eight sentences were produced in each condition. Plosive-vowel syllables were selected for measurement before the color words, at the color words, and after the color words. All participants exhibited longer and more extensive vowel transitions before and at the color word in comparison to afterwards. In the incongruent condition, the color words had longer F2 transition durations. The older adults had longer F2 transition durations than the younger adults. The F2 transition frequency extent was greatest at the color word. The older adults had wider F2 transitions at the color word that also were wider for the incongruent words.

5aSC25. Imitation of prosodic contours in word shadowing. Cynthia G. Clopper (Ohio State Univ., 1712 Neil Ave., Oxley Hall 100, Columbus, OH 43210, clopper.1@osu.edu)

In a word shadowing task, participants are asked to repeat auditorily-presented words aloud, providing a speech "shadow" to the stimulus materials. Previous research has demonstrated that even in this kind of non-interactive, non-social task, participants' speech is more similar to the stimulus materials during shadowing than during a baseline reading task. These results are taken as evidence for implicit phonetic imitation during word shadowing. The current study examined imitation of prosodic contours in a word shadowing task. Participants were asked to first read a set of words aloud to establish their baseline production and then to repeat the same set of words after hearing them produced by a young female native speaker of Midwestern American English. This model talkers' word productions all involved a non-falling intonation contour (i.e., either a plateau or a rise), consistent with "list intonation" in reading. Although the participants produced more than 50% non-falling contours in their baseline productions, they produced significantly more non-falling contours in the shadowing task, suggesting imitation of the prosodic contours produced by the model talker. The results of an acoustic analysis of the prosodic contours to examine the phonetic detail of the imitation will also be discussed.

5aSC26. Individual strategies in adaptation to altered auditory feedback. Sarah Bakst (Commun. Sci. and Disord., Univ. of Wisconsin - Madison, 1500 Highland Ave., Madison, WI 53705, sbakst@wisc.edu), John F. Houde (Otolaryngol. - Head and Neck Surgery, UCSF, San Francisco, CA), Susan Lin, and Keith Johnson (Linguist, Univ. of California Berkeley, Berkeley, CA)

Speakers monitor themselves while talking. When they hear a real-time altered version of their speech, they will change their articulation so that when they hear their altered speech, it matches their acoustic target [Houde and Jordan (1998, Science 20;279(5354):1213–1216)]. The experiment presented here used the novel addition of ultrasound imaging to reveal how speakers ($n = 30$) change their articulations in response to two different formant perturbations: raising of F1 in "head" and F2 in "hood." Principal components analysis was used to identify speakers' individual strategies during adaptation. Some speakers use a single strategy for an entire adaptation block, while others change strategy. Speakers are also known to change production in a formant that was not altered [Katseff *et al.* (2010, JASA **127**(3), 1955)]. The ultrasound analysis shows that at least for some speakers, change in two formants is linked to independent and uncorrelated articulatory components and possibly serves a perceptual purpose in compensation, rather than being an unintended result of the compensatory response. Preliminary results ($n = 4$) of adaptation to raising F3 in "heard" will also be presented. Modeling with the Maeda and Manzara synthesizers [Bakst and Johnson (2016, JASA **140**(4), 3223)] correctly predicted speakers' articulatory strategies.

5aSC27. Acoustic cues to linguistic profiling? Machine learning of phonetic features of African American English. Meisam K. Arjmandi, Laura Dilley, and Suzanne Wagner (Dept. of Communicative Sci., Michigan State Univ., East Lansing, MI 48824, ldilley@msu.edu)

Listeners can readily differentiate words spoken in an African American English (AAE) dialect from a Standard American English (SAE) dialect, even in absence of distinctive morphosyntactic features. However, it is still unclear what acoustic-phonetic cues listeners utilize to rapidly distinguish AAE from SAE. This study investigates the informativeness of various acoustic-phonetic cues to the characterization of AAE dialect. V and VC sequences (with C = /n/, /m/, /l/, /r/) from speech of 7 female speakers (4 SAE and 3 AAE), recorded during sociolinguistic interviews, were randomly selected and acoustically analyzed, controlling for coarticulatory context. Acoustic cues of F1, F2, F3, F4 formant trajectories, formant bandwidth, pitch variation, duration, intensity, and voice quality measures (e.g., harmonic-to-noise ratio, jitter, shimmer, and spectral slope) were measured in these segments to identify the extent of their contribution to separating AAE and SAE. The results from machine learning modeling of acoustic cues demonstrate that speech within an AAE dialect entails distinct acoustic characteristics and voice quality compared to SAE speech. These separate acoustic patterns between AAE and SAE dialect indicate the need for including dialect-specific acoustic cues both in automatic speech recognition applications and clinical assessments of speech-language disorders.

5aSC28. Modeling geographic variation in pronunciation of United Kingdom English. Katherine Henley (Linguist, Univ. of Georgia, 142 Gilbert Hall, Athens, GA 30602, keh71529@uga.edu) and John S. Coleman (Linguist, Philology and Phonet., Univ. of Oxford, Oxford, United Kingdom)

This paper details an investigation of geographic variation of UK English pronunciation based on formant trajectories of the diphthongs /aɪ/ and /aʊ/. Thousands of audio samples of these vowels were retrieved from the spoken portion of the British National Corpus (BNC), and their F1 and F2 values were compared across 89 locations within the UK. The study was designed with an expectation of a spectrum of pronunciations existing across the UK with monophthongal realizations primarily found in the Northern and Western areas of the country and more diphthongization occurring in the South. After searching the BNC for phonemically aligned tokens of "five" and "house," functional data, modal, and geographic analyses were conducted. We found a strong correlation between speaker distance from the dataset's northernmost location and diphthongization of /aɪ/. This finding suggests that the variation in pronunciation of the vowel in "five" is more continuous than isoglossic. By investigating formant trajectories of vowels involved in the English Great Vowel Shift, we have captured variation in pronunciation that exists across the UK as a result of English sound change.

5aSC29. Shifting from the shift: Loss of dialect distinction in the US. Monica Nesbitt (Linguist and Lang., Michigan State Univ., Wells Hall - MSU, 619 Red Cedar Rd., East Lansing, MI 48824, nesbit17@msu.edu)

Historically, /æ/ in the Northern Cities Shift (NCS) dialect area is realized as raised, lengthened, and diphthongized in all consonantal contexts (Boberg and Strassel 2000) and that pre-oral /æ/ is realized with as much nasalization as pre-nasal /æ/ (Plichta 2004). Recent studies suggest that young speakers in the dialect area are adopting a nasal pattern for /æ/ such that only tokens before nasal consonants are raised and fronted in phonetic space (in Lansing, MI (Wagner *et al*. 2016), in Syracuse, NY (Driscoll and Lape 2015), in Upstate NY (Thiel and Dinkin 2017), in Detroit, MI (Acton *et al*. 2017)). Through acoustic analysis of 1310 /æ/ tokens produced by 26 speakers born and raised in Lansing, MI (date of birth 1991–1997), the current study finds that younger speakers are not simply lowering and retracting this vowel, they are rejecting all phonetic components of the NCS /æ/ system. For these speakers, only pre-nasal /æ/ tokens are realized as long, diphthongal, and nasalized, while pre-oral tokens are short, monophthongal, and have little to no nasalization. Vowel quality for /æ/ in the NCS dialect area is thus undistinguishable from that in the western and midland states in the US (e.g., California and Kansas).

5aSC30. Using speech stereotypes to assert identity and affinity with a minority group. Auburn Lutzross (Linguist, Univ. of California Berkeley, 2435 Grant St., Apt. 2, Berkeley, CA 94703, alutzross@berkeley.edu)

Perception-based studies on female speech have demonstrated clear stereotypes associating acoustic qualities and sexual orientation (e.g., Camp 2009, Munson *et al*. 2005, Pierrehumbert *et al*. 2004). However, production studies on female speech are rare and inconclusive: though male speakers have been shown to vary their use of stereotypically gay speech according to context (Podesva 2011), female speakers have not been studied. Likewise, there has been little consideration of how other personal attributes can intersect with sexual orientation to affect use of stereotypical speech. This study addresses the following for female speakers: (1) how personal attributes interact with sexual orientation in phonetic variation, (2) why straight or bisexual speakers might take on features of stereotypically lesbian speech, and (3) how these personal factors affect speech in different contexts and the resulting inter-speaker and inter-context variation. Speakers were recorded in both reading and interview speech modes. ANOVA of phonetic variation revealed that for straight and bisexual speakers, their "familiarity with Queer culture" was the most influential attribute on both speech modes. However, there was no such effect for lesbian speakers. I argue that speakers used stereotypical lesbian speech patterns to express outgroup affinity.

5aSC31. A sociophonetic analysis of gay male speech stereotypes in Buenos Aires, Argentina. Ellis Davenport (Linguist, Macalester College, 1600 Grand Ave., St. Paul, MN 55105, edavenpo@macalester.edu)

The present study expands academic knowledge on the phonetic cues that index "gay speech" beyond the English language by examining speech produced by gay Buenos Aires Spanish (porteño) speakers. Research has been conducted on gay speech in other parts of Latin America or on the Spanish language in general (Pahis 2017, Ezquerra 2015, Mack 2011, Mendes 2007, Sívori 2005), but not on the porteño dialect. Acoustic features (vowel quality, pitch, speech rate, and degree of aspiration of pre-consonantal and syllable-final /s/) corresponding to public stereotypes on porteño gay speech were measured. Speech from gay and straight men, as well as actors interpreting gay & straight roles in film and television, was analyzed. It is hypothesized that gay men will have higher pitch and that vowels be more clearly articulated than straight counterparts, that pre-consonantal and syllable-final /s/ will be aspirated less frequently and that pitch contours will be more pronounced in the speech of gay men. In addition, the speech of actors playing gay roles will align more closely with the stereotypes of gay speech than with actual phonetic properties, in accordance with Cartei and Reby 2011.

5aSC32. Understanding the speech cues to bisexuals. Mariya Yoshovska (Linguist, Macalester College, 1600 Grand Ave., St. Paul, MN 55105, myoshovs@macalester.edu)

The vast majority of research on speech patterns in the queer community has focused on gay and lesbian speakers. This study aims to expand our knowledge of speech in the queer community by focusing on the speech cues of adult bisexual speakers. For this study, adult lesbian, gay, bisexual, and straight cisgender American English speakers recorded the "Fire passage," the "Rainbow passage," and a description of their hometown. Vocal characteristics, such as duration, pitch, voice quality, formants, and speech rate, were measured to observe the differences between speech patterns of adult bisexual speakers and other queer and straight identities. The results suggest that bisexual and lesbian female speakers have naturally lower average pitch while straight speakers go up in pitch. Bisexual speakers maintain their pitch range between natural and read speech, while lesbian speakers have a wider pitch range during natural speech and straight speakers during read speech. The speech rate of bisexual and straight speakers decreases in natural speech, while it increases for lesbian speakers. Sentence-final phonation results were not conclusive between speaker identities. It is hypothesized that bisexual speakers produce less fronted /u/ and /ɑ/, higher /ɛ/ (low F1) and a more back /o/ (low F2).

5aSC33. Growth in the accuracy of preschool children's /r/ production: Evidence from a Longitudinal Study. Mara Logerquist (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), Hyuna A. Kim (Commun. Sci. and Disord., Univ. of Wisconsin, Madison, WI), Alisha B. Martell, Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu), and Jan Edwards (Hearing and Speech Sci., Univ. of Maryland, College Park, MD)

The acquisition of American English /r/ is of particular theoretical and clinical interest because it is typologically rare among the world's languages, because it is one of the latest-acquired sounds by typically developing children, and because it is one of the sounds that is most likely produced in error by children with residual speech sound disorders. In this study, we examined the accuracy productions of /r/ and its most frequent substitute, /w/, by preschool children (n = 120) at two time-points: when they were between 3;3 (years;months) and 4;4 years old and again when they were 4;4 to 5;4 years. Accuracy was determined both by trained phonetic transcriber. Moreover, the accuracy of productions from the first time-point was rated by groups of naive listeners using a perceptual rating experiment similar to that described by Schellinger *et al*. (2016). Considerable variation in /r/ production was found at both time-points, and there was sizeable variation in the extent to which production improved from the first to the second time-point. Surprisingly, none of the predictor measures at the first time-point (speech perception, expressive and receptive vocabulary size) predicted growth in /r/ in this group.

5aSC34. Towards an inventory of prosodic contours produced by English-acquiring 2-year-olds: The use of H* and H + !H*. Jill C. Thorson (Commun. Sci. and Disord., Univ. of New Hampshire, 4 Library Way, Hewitt Hall, Dover, NH 03824, jill.thorson@unh.edu) and Stefanie Shattuck-Hufnagel (Speech Commun. Group, RLE, MIT, Cambridge, MA)

One viewpoint of early prosody is that utterances are highly simplified in comparison to adult models, in part, due to biological constraints such as incomplete control of pitch production (Lieberman, 1967; Snow, 2006). In contrast, research in Catalan and Spanish shows that early utterances (under 2 years) consist of the basic intonational categories (Prieto *et al*., 2012), including contours that would not be predicted under a biological approach. Our primary question asks what is the inventory of contours produced by American English-acquiring toddlers. Based on work showing toddlers to have near adult-like acoustic realizations of certain pitch accents (Thorson, 2015), we hypothesize that children will demonstrate sophisticated intonational configurations; here we analyze specific pitch accent varieties in

toddler speech. Spontaneous utterances of four 2-year-olds were analyzed. Pitch accents were annotated using ToBI (Beckman & Ayers, 1997) and acoustically analyzed. Results reveal three varieties of high pitch accents: (1) H*, rise in f0 on stressed syllable; (2) H + !H* complex-bitonal, higher f0 on preceding syllable, falling f0 throughout accented syllable; (3) H + !H* type-2, f0 plateau on preceding syllable continuing onto stressed syllable. Examining how toddlers use specific intonational elements helps to identify the prosodic contours that make up the child inventory.

5aSC35. Variation in English infant-directed speech. Isabelle Lin (Dept. of Linguist, Univ. of California Los Angeles, 3125 Campbell Hall, UCLA, Los Angeles, CA 90095-1543, isabellelin@ucla.edu), Adam J. Chong (Linguist, Queen Mary Univ. of London, London, United Kingdom), and Megha Sundara (Linguist, Univ. of California Los Angeles, Los Angeles, CA)

In adult-directed speech (ADS), words are rarely produced canonically. Infant-directed speech (IDS) has been argued to contain more canonical productions. However, recent analyses show that IDS is as variable as ADS. Then, how could infants learn to privilege canonical forms, as has been shown for adult listeners? Previous research on variation in IDS has focused on word-final productions. In this study, we investigate whether the extent of variation in IDS differs by the position of a segment in a word. We sampled IDS to 6 infants between 16 and 24-mo-old from the Providence corpus. Utterances with /t/, /d/, /n/, /s/ and /z/ were identified orthographically, forced-aligned, corrected, and transcribed by 3 phonetically-trained native speakers of English. This yielded 28,775 segment tokens in word initial, medial and final position. Results confirmed that IDS is at least as variable as ADS (canonical pronunciations $< 50\%$). However, variation was limited to coda positions; on average, over 90% of onsets were produced canonically compared to just 60% of codas. This positional difference could benefit category learning. Word-initial segments would bolster acquisition of canonical forms, and possibly support word segmentation, while word-final variation would encourage learning of phonetic variants resulting from processes in connected speech.

5aSC36. The influence of siblings on toddlers' mean length of utterance. Mark VanDam, Allison Saur (Speech & Hearing Sci., Washington State Univ., PO Box 1495, Spokane, WA 99202, mark.vandam@wsu.edu), Jenna Anderst (Com Sci Disord, East Wash Univ, Spokane, WA), Daniel Olds (Speech & Hearing Sci., Washington State Univ., Spokane, WA), and Paul De Palma (Comput. Sci., Gonzaga Univ., Spokane, WA)

Linguistic complexity is an indicator of language development in young children. Complexity of a child's linguistic productions have been shown to increase with development, but may be affected by factors such as disability or environmental variables. Here, we look into the role of family composition as a possible influence on a child's developing ability to use increasingly complex language. In particular, we ask if a toddler's mean length of utterance (MLU) is affected by the presence of siblings in the family and whether the sex of the child may play a role. MLU values were extracted from the public HomeBank database [http://homebank.talkbank.org] of transcribed natural child speech for both the target toddler and for siblings present in the recordings. Results indicate a main effect of increased MLU in children without siblings, but interaction effects suggest that differences may be driven by the boys without siblings alone. There was no correlation between the MLU of the target child and the MLU of the sibling. Findings are discussed in terms of family dynamics and joint attention.

5aSC37. Changes in vowel space characteristics during speech development based on longitudinal of measurements of formant frequencies. Brad H. Story, Kate Bunton, and Rebekkah Diamond (Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., P.O. Box 210071, Tucson, AZ 85721, bstory@email.arizona.edu)

During speech development, a child's vocal tract undergoes changes due to growth of anatomic structures. Such changes typically lower the formant frequencies, reshaping the [F_1,F_2] vowel space. Much of what is known about vowel space change, however, is based on cross-sectional formant measurements averaged over children in various age groups. The purpose of this study was to characterize changes in the vowel space of four children between the ages of 2 and 6 years. Longitudinally-collected audio recordings of four children (2F,2M) were selected from the Arizona Child Acoustic Database. Each child had been recorded every four months from ages 2-6 years, and produced a variety of words, phrases, vowel-vowel progressions, and occasional spontaneous speech. Formant frequencies (F_1 and F_2 only) were measured from the recordings using a spectral filtering technique. At each age increment, the formant frequencies for each child were plotted as vowel space density, where the "density" dimension indicates the relative tendency of a talker to produce sound in particular region of the vowel space. The change in location and shape of the density cloud during this period of development will be demonstrated. [Research supported by NIH R01-DC011275, NSF BCS-1145011.]

Session 5aSP

Signal Processing in Acoustics and Underwater Acoustics: Continuous Active Acoustics

Zachary J. Waters, Chair
Physical Acoustics - Code 7130, Naval Research Laboratory, 4555 Overlook Ave. SW, Bldg 2. Rm. 186, Washington, DC 20375

Invited Papers

8:00

5aSP1. Towards Doppler estimation and false alarm rejection for Continuous Active Sonar. Jeffrey R. Bates (Ctr. for Maritime Res. and Experimentation (NATO-STO), STO-CMRE, Viale San Bartolomeo 400, La Spezia, SP 19126, Italy, jeffrey.bates@cmre.nato.int), Doug Grimmett (SPAWAR Systems Ctr. Pacific, San Diego, CA), Gaetano Canepa, and Alessandra Tesei (Ctr. for Maritime Res. and Experimentation (NATO-STO), La Spezia, SP, Italy)

Linear frequency modulated (LFM) continuous active sonar (CAS) waveforms show promise for use in target tracking given that waveforms can be split into sub-waveforms (sub-bands), thereby increasing the target refresh rate. However, reducing the duration and bandwidth of the sub-band decreases the SNR/SRR, adversely affecting detection. We present a target detection technique in which sub-bands are averaged incoherently while exploiting the range bias error (commonly observed in large duration LFM CAS waveforms with significant Doppler) to significantly improve detection. Sub-band averaging, known to reduce the false alarm rate, also mitigates channel coherence losses while maintaining detectability in CAS. A method for performing incoherent averaging over many possible Doppler shifts and identifying contacts via clustering in the 3-dimensional range-bearing-Doppler parameter space will be described. Finally, the promising results obtained with this technique on data acquired by the 2016 Littoral CAS Multi-National Joint Research Project sea trial will be shown. [This work was funded by NATO Allied Command Transformation Future Solutions Branch under the Autonomous Security Network Programme and the LCAS MN-JRP.]

8:20

5aSP2. On the use of decision feedback equalization for continuous active sonar. Konstantinos Pelekanakis, Jeffrey R. Bates, and Alessandra Tesei (Res. Dept., Ctr. for Maritime Res. and Experimentation (NATO-STO), NATO STO CMRE, Viale San Bartolomeo 400, La Spezia, SP 19126, Italy, jeffrey.bates@cmre.nato.int)

Continuous Active SONAR (CAS) systems allow duty cycles up to 100% and so lower target association errors are possible as compared to Pulse Active Sonar (PAS). Large time-bandwidth product Linear Frequency Modulated (LFM) signals are the de facto standard for this type of SONAR systems, yet, these signals suffer from low processing gains when the ocean exhibits low-spatio temporal coherence. In this work, we depart from the typical sub-band matched filter processing and propose to adopt signal processing techniques used in underwater acoustic communications. In particular, we analyze Binary Phase Shift Keying (BPSK) signals via an adaptive Decision-Feedback Equaliser (DFE). The DFE is able to adapt to environmental changes based on the known transmitted bits. The operational bandwidth is 1500–4000 Hz and the BPSK signals are transmitted in two non-overlapping bands to avoid transmit interference during reception. The performance of the proposed system is demonstrated based on field data recorded during a sea experiment off the Coast of La Spezia in October 2017. The key result here is that the Doppler measurement update rate of the echoes is as fast as the bit rate of the transmitted BPSK signal.

8:40

5aSP3. Parametric bandwidth synthesis for active sonar. Jonathan B. Soli (Elec. and Comput. Eng., Duke Univ., 141 Providence Rd., Chapel Hill, NC 27514, jonsoli5@gmail.com) and Jeffrey Krolik (Elec. and Comput. Eng., Duke Univ., Durham, NC)

This talk presents an unconventional range processing technique applicable to linear FM pulsed and continuous active sonar. Parametric bandwidth synthesis (BWS) involves performing linear-predictive extrapolation using a low-order parametric autoregressive (AR) model to extend the bandwidth of received sonar returns. Synthetically increasing the bandwidth of echoes results in improved range resolution and greater pulse compression gain compared to standard matched filtering. It should be noted that despite the resolution improvement, BWS does not improve ranging accuracy beyond that of the transmitted signal. Using a low-order AR model for BWS enhances specular target returns and discrete scatterers, while not enhancing diffuse multipath, reverberation, and noise returns. BWS performance was verified in simulation and with sea trial data from the Littoral Continuous Active Sonar 2015 (LCAS-15) experiment. Processed results confirm that BWS improves range resolution of the echo repeater target, while suppressing multipath, diffuse reverberation, and noise. Analysis of the practical limits of BWS are also included.

9:00

5aSP4. Waveform performance and sidelobe reduction techniques for continuous active sonar (CAS). Matthew T. Adams, Brian B. Bishop, and Nicholas A. Rotker (MITRE Corp., 202 Burlington Rd., Mailstop S118, Bedford, MA 01730, mtadams@mitre.org)

The ability to conduct surveillance and tracking of targets over a wide area necessitates the use of a distributed sensing network and an appropriate processing scheme. One such technique that may be appropriate for the target tracking problem is continuous active sonar (CAS). In September 2016, an experiment was performed in Narragansett Bay, RI, to assess the performance of CAS for tracking a single unmanned underwater vehicle (UUV). PN sequence-coded chirps and LFM pulses of various bandwidths and center frequencies were transmitted from one transducer and received at two hydrophones in different locations. Following the experiment, several range-doppler sidelobe reduction techniques were experimented with to improve target detections and range-doppler estimations. During processing, waveform properties critical to sidelobe reduction performance were identified, and new waveforms have been chosen which exhibit these properties. A new experiment will be conducted in the coming months to demonstrate the performance of these waveforms, and results are expected to show improved detections and range-doppler estimations for various targets.

9:20

5aSP5. Demonstration of a compact quasi-monostatic autonomous underwater vehicle based continuous active sonar. Zachary J. Waters (Physical Acoust. - Code 7130, Naval Res. Lab., 4555 Overlook Ave. SW, Bldg 2. Rm. 186, Washington, DC 20375, zachary.waters@nrl.navy.mil)

Monostatic active sonar systems are typically operated in a pulsed transmit configuration, where an acoustic source is first activated to provide ensonification of the environment and then de-activated while scattered and ambient environmental acoustic energy is collected by a receiving sensor. In applications where the transmitter and receiver are in close proximity as with a monostatic configuration, it is typically impracticable to carry out the simultaneous transmission and reception of acoustic energy. Under such conditions, during continuous transmission, the source may saturate the receive sensors, potentially resulting in an overlap of target scattering returns from down-range with the direct acoustic coupling response from the source-to-receiver as well as reverberation, effectively blinding the sonar. Here, we explore the feasibility of techniques to disambiguate lower level target returns from a continuously transmitting autonomous underwater vehicle (AUV) based sonar. We conclude via theoretical studies supported by empirical analysis that the continuous transmission and reception of acoustic energy in a shallow-water waveguide is feasible for a realistic AUV based sonar system and overview demonstrations of these techniques to detect a bottomed target-object at-sea. [Work Supported by the Office of Naval Research.]

9:40

5aSP6. Inference regarding the state of a mobile underwater object from narrow band observations on a small aperture vertical array. Abner C. Barros (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Darmouth, MA 02747, abarros1@umassd.edu) and Paul J. Gendron (ECE Dept., Univ. of Massachusetts Dartmouth, North Dartmouth, MA)

A narrowband source ensonifies an area of interest in a refractive undersea environment. A small aperture vertical array is employed to infer the depth, range and speed of the acoustic scatterer by resolving the scattered direct and surface interacting wave fronts. Tracking the target by means of a continuous wave transmission is challenging due to the difficulty of inferring the frequencies and angles of the two returned closely spaced wave vectors. The propagation of sound in a refractive medium presents additional challenges for inversion of the wave vectors to range, depth, and speed. Joint posterior inference is made possible with a computational Bayesian Gibbs sampling scheme over all track parameters taking full advantage of the analytic tractability of the conditional densities of the received amplitudes and phases and of the ambient acoustic noise power. The conditional densities of the ordered wave vectors however are constructed numerically by 2-dimensional inverse quantile sampling. The inferred joint posterior density of the target state is obtained by constructing a numerical inverse transformation of the acoustic propagation model and provides posterior confidence intervals over speed, depth, and range. Simulation results demonstrate the approach at received signal to noise ratios (SNR) well below -9 dB and illuminate the limits of depth and speed estimation as a function of both depth and SNR.

Session 5aUWa

Underwater Acoustics: Underwater Soundscape: Measurement and Characterization

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

Contributed Papers

8:00

5aUWa1. Field verification of acoustic sources of geophysical survey in shallow water conditions. Sei-Him Cheong and Laura Palmer (Marine Wildlife Dept., Gardline Geosurvey Ltd., Endeavour House, Admiralty Rd., Great Yarmouth, Norfolk NR30 3NG, United Kingdom, sei-him.cheong@gardline.com)

Underwater noise generated from offshore survey is a growing concern and is known to have ecological consequences on the marine environment. As a conservation effort to reduce anthropogenic impact to the marine ecosystem, BOEM requires acoustic field verifications of sound sources used in a geophysical survey conducted in US water, to ensure the sound fields will not have detrimental effect to marine wildlife. This paper presents the field verification data from two geophysical surveys conducted in the east coast of USA in summer 2016. The survey locations were considered as area of high ecologically importance due to the close proximity to the migratory routes for a number of Baleen whale species including Fin and Northern Right whale. The recording of the survey sources were analysed spatially based on their spectral and temporal characteristics, to quantify the noise exposure experienced by a potential receptor. A comparative assessment between the sound isopleths obtained from field measurement and practical mitigation area was conducted. This information not only gives invaluable insight on the acoustic propagation in the local environment; it also provides extra confidence for the mitigation practice throughout the geophysical survey.

8:15

5aUWa2. Acoustic ground truthing of seismic noise in Chatham Rise, New Zealand. Sei-Him Cheong and Breanna Evans (Marine Wildlife Dept., Gardline Geosurvey Ltd., Endeavour House, Admiralty Rd., Great Yarmouth, Norfolk NR30 3NG, United Kingdom, sei-him.cheong@gardline.com)

Noise generated by seismic survey is widely recognised as a pervasive pollutant to marine ecosystem. Between the 31st January and 21st March 2016, a geophysical research survey was conducted in Chatham Rise, New Zealand, to collect seismo-acoustic data using a Sercel seismic streamer in order to ground-truth the underwater noise impact assessment, conducted according to the DOC (NZ) Seismic Survey Code of Conduct. Data were analyzed to determine the received sound level at a distance up to 3 km from the source array. This paper establishes the method to predict the impact radii in order to validate the results obtained using Gardline 360M predictive model. The aim was to provide confidence to the capability of predictive modelling for estimating the impact zone of a seismic sound source. Data showed that multipath reflections can fluctuate significantly according to the seafloor topography; however, a very consistent trend can be obtained from direct propagation to confidently establish mitigation radii. Results show that the employment of a seismic streamer for the establishment of effective mitigation radii is technically feasible and may be used as a tool to ground truth predictive modelling as part of mitigation plans to reduce the potential risk of acoustic trauma.

8:30

5aUWa3. The noise of rock n'roll: Incidental noise characterization of underwater rock placement. Rute Portugal, Sei-Him Cheong, Breanna Evans (Marine Wildlife Dept., Gardline GeosurveyLtd., Endeavour House, Admiralty Rd., Great Yarmouth, Norfolk NR30 3NG, United Kingdom, rute.portugal@gardline.com), and James Brocklehurst (Boskalis, Papendrecht, Netherlands)

Underwater noise is a growing concern to conservation and stock management efforts to which supra-national organizations (e.g., OSPAR or the European Union) and governments (e.g., USA) are beginning to respond by building catalogs of the noise introduced in the marine environment by human activity. Rock placement is a construction activity for which there is scarcely any data available. In order to fill the knowledge gap, opportunistic recordings were taken while the Gardline Mk 3 hydrophone array was deployed for Passive Acoustic Monitoring and mitigation for marine mammals. The recordings were analyzed for their spectral and temporal characteristics, a correlation analysis between the amount of rock placed and the intensity of sound produced was made and the suitability of the hydrophone array for the collection of this type of data was assessed.

8:45

5aUWa4. Study on single raindrop noise and energy conversion. Dajing Shang, Qi Li, Rui Tang, Fangzhou Deng, and Shu Liu (Underwater Acoust. Eng. Dept., Harbin Eng. Univ., Nangang District Nantong St. No.145, Harbin City, Heilongjiang Province 150001, China, shangdajing@hrbeu.edu.cn)

The measurement of rainfall in the ocean is more difficult than in land, but the noise signals produced by rainfall can be used for measuring the rainfall in the ocean. In this paper, a single raindrop measurement system was set up to measure the noise of the raindrop, the bubble sound are classified by the radius of the raindrop which was measured by the splash method. The mechanism of the initial impact and the bubble sound are analyzed, and the kinetic energy threshold and acoustic energy conversion efficiency of the single raindrops are also investigated. The results show that the bubble noise is the main noise compared with initial impact noise, the initial impact noise will become larger with the radius of raindrop, the acoustic energy of great and large raindrop is much larger than that of small one, but the tiny and medium raindrop have no acoustic energy. The kinetic energy threshold is not a constant, but it is proportional to the raindrop size.The acoustic energy conversion rate is about 1.04×10^{-3}% for small raindrops, 10^{-5}% for heavy rains and great raindrops, there are almost no acoustic energy for tiny raindrops and medium-sized raindrops.

Session 5aUWb

Underwater Acoustics: Underwater Acoustic Propagation: Models and Experimental Data

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

Contributed Papers

9:30

5aUWb1. Integrating the Biot-Stoll model with contact squirt flow and shear drag (BICSQS) in the Biot viscosity extended framework. Sri Nivas Chandrasekaran, Karen Brastad Evensen, Elisabeth Grønn Ramsdal, and Sverre Holm (Dept. of Informatics, Univ. of Oslo, Gaustadalléen 23B, Oslo 0373, Norway, srinc@ifi.uio.no)

The viscosity extended Biot theory [Sahay, Geophysics (2008)] incorporates pore fluid viscosity in the constitutive relation of the classical Biot theory. The strain rate term due to pore fluid viscosity turns the zero velocity slow S wave mode into a diffusive process. In this framework, it is claimed that fast P and S waves bleed energy when the slow S wave is generated at an interface or at a discontinuity. On the other hand, the Biot-Stoll model with contact squirt flow and shear drag (BICSQS) [Chotiros and Isakson, JASA (2004)] models the grain to grain contact as compression and shear relaxation processes and obtains a causal poro-elastic frame response. In this work, we incorporated the grain to grain contact physics in the constitutive relation of the Biot theory and integrated it into the viscosity extended Biot framework. With realistic parameters, the fast S wave phase velocity and attenuation in the integrated model show a significant difference over a range of frequencies when compared to BICSQS. Additionally, this integrated model may provide a more accurate match for the measured reflection coefficient.

9:45

5aUWb2. Sound propagation effects of near-seabed internal waves in shallow water. Timothy F. Duda, Andone C. Lavery, Ying-Tsong Lin, and Weifeng G. Zhang (Woods Hole Oceanographic Inst., WHOI AOPE Dept. MS 11, Woods Hole, MA 02543, tduda@whoi.edu)

Computational modeling is used to examine effects of near-seabed nonlinear internal waves. Waves of this type have been observed many times on the South New-England Shelf and elsewhere. The waves appear to be common when the dominant vertical density gradient zone is near the seabed. Many observations have been with profilers that alias the waves, or with instruments fixed to the seafloor that provide snapshots of passing waves. We have recently observed these features in greater detail using acoustic remote sensing from a ship, and we are now in a position to model their effects on sound propagation. Results are presented for many frequencies and many source/receiver/wave geometries. An important question remains about along-crest coherence of the waves, which would govern ducting and other anisotropic propagation effects. These waves are not as well studied as waves linked to a near-surface pycnocline, which can be studied remotely with EM via surface signature. The strong density gradients found in many shallow water areas allow the internal waves to have high frequencies, creating rapidly changing acoustics thereby shortening acoustic coherence time. The related strong sound-speed gradients, found both near the surface and near the seabed, are responsible for the important acoustic effects of the waves.

10:00

5aUWb3. Normal mode coupling through horizontally variable sound speed. Cathy Ann Clark (Sensors & Sonar Systems, NUWCDIVNPT, 1176 Howell St., B1320, R457, Newport, RI 02841, cathy.clark@navy.mil)

In conjunction with construction of a normal mode model specifically designed to compute acoustic propagation in shallow water environments, the coupling associated with horizontally variable sound speed is predicted using a closed form solution in a modified Pekeris wave guide with a hard bottom. The code enables analysis of the correspondence between mode summation and the corresponding wave propagation as represented by ray propagation. The closed form solution is being used to provide insights into horizontal mode coupling and as a benchmark for verification of coupling in the shallow water normal mode solution.

10:15

5aUWb4. Impact of salinity on predicted transmission loss. Jennifer Cooper and Michael Mandelberg (Appl. Phys. Lab., John Hopkins Univ., 11100 Johns Hopkins Rd., Mailstop 8-220, Laurel, MD 20723, jennifer.cooper@jhuapl.edu)

Often ocean sound speed profiles are estimated using a temperature profile measured, for example, with an expendable bathythermograph (XBT) combined with a database value for salinity. The prevalence of this approach is due to the added expense and limited availability of sensors that can simultaneously measure both temperature and salinity profiles, for example conductivity-temperature-depth (CTD) probes. When predicting acoustic performance of a sonar system, using the sound speed profile based only on measured temperature is not always sufficient. This study examines when and where the use of temperature-only profiles is adequate as well as potential improvements to typical sound speed profiles that can be made using limited conductivity measurements (i.e. measurement at a single depth or a small segment of the profile) or with database values based on oceanographic models or chosen based on matching the temperature profile rather than the geographically nearest value. The assessment of a profile's adequacy is based on how well modeled transmission loss (TL) matches the TL modeled using the measured temperature and salinity profile from CTD.

10:30–10:30 Break

10:30

5aUWb5. An immersed interface method for the solution of the wide-angle parabolic equation in range-dependent ocean environments. Roberto Sabatini (CNRS-LMA, Ecully cedex, France) and Paul Cristini (CNRS-LMA, 4, Impasse Nikola Tesla, CS40006, Marseille 13013, France, cristini@lma.cnrs-mrs.fr)

Taking into account accurately a complex topography, within the framework of the parabolic equation method in underwater acoustics, is a difficult

task. Several methods have been successfully proposed in the literature in order to improve the accuracy of numerical simulations. Nevertheless, despite the recent improvements, finding an accurate solution remains an open problem. In this presentation, we propose a novel approach for the treatment of irregular ocean bottoms within the framework of the wide-angle parabolic equation. This new approach is based on the immersed interface method originally developed by LeVeque and Li [SIAM J. Numer. Anal. 31(4), 1019–1044, (1994)]. It is intrinsically energy-conserving and allows to naturally handle generic range-dependent bathymetries. As an illustration of its capabilities, we provide some promising results based on the solution of different benchmark problems.

10:45

5aUWb6. Modeling acoustic wave propagation and reverberation in an ice covered environment using finite element analysis. Blake Simon and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 1400 Briarcliff Blvd., Austin, TX 78723, blakesimon8@gmail.com)

A three-dimensional, longitudinally invariant finite element model of acoustic propagation and reverberation in an ice-covered shallow water waveguide has been developed. The ice is modeled as both an elastic medium and a pressure release surface. Transmission loss levels are calculated and compared for both assumptions of ice. Using Fourier synthesis, the time-harmonic acoustic pressure results are transformed into the time domain, and reverberation levels are then compared for both models. Finally, using a fully three-dimensional version of the finite element model, compressional-to-shear wave conversion at the elastic ice and water interface is characterized to inform propagation mechanisms of acoustic waves in Arctic ice sheets. [Work supported by ONR, Ocean Acoustics and the Robert W. Young Award for Undergraduate Student Research in Acoustics.]

11:00

5aUWb7. A characterization of the instantaneous wave front distortion characteristics in SW06 and ASIAEX 2001 data and strategies for exploiting lucky scintillations. Ivars P. Kirsteins (NUWCDIVNPT, 1176 Howell St, Newport, RI 02841, i.kirsteins@gmail.com)

Long line arrays are highly susceptible to signal wave front distortions caused by random medium effects like internal waves. Effects include large localization errors, biases, and loss of array gain. In earlier work, we had showed evidence from real data [Kirsteins and Ge, IEEE UASP workshop, Oct. 2017] that lucky scintillations, i.e., moments when the instantaneous signal wave front is relatively undistorted, occur regularly at much shorter time scales even during periods of strong internal wave activity that could potentially be exploited to improve array processing in environments with apparently poor spatial coherence. To better understand the phenomenon of lucky scintillations and how they can be used in array processing, we characterize in this paper the instantaneous or short time signal wave front distortion behaviors of horizontal line array (HLA) data from the Shallow Water 2006 (SW06) and ASIAEX 2001 experiments provided by the Woods Hole Oceanographic Institute, examining the time scales and the rate at which these lucky scintillations occur. Our analysis suggests an alternative array processing strategy by collecting data snap shots over a time interval matched to the ocean random medium time scales and using only the best snap shots in the estimator.

11:15

5aUWb8. Numerical study on radiation impedance of underwater sound sources in non-anechoic tank. Yiming Zhang, Rui Tang (Harbin Eng. Univ., Nantong St., Nangang District No. 145, Harbin 150001, China, zhangyiming1993@hrbeu.edu.cn), Qi Li, and Xin Feng (Harbin Eng. Univ., Harbin, Heilongjiang, China)

The radiation impedance is an important index to evaluate the characteristics of underwater sound sources. In this study, a numerical method for calculating the radiation impedance of underwater sound sources using ACTRAN is proposed. The radiation impedance is calculated by extracting the sound pressure and velocity on the surface of the sound sources. To verify the accuracy of the numerical method, the radiation impedance of a pulsating sphere source is calculated and the results are in good agreement with the analytical results. The radiation impedance of an elastic spherical shell in air, anechoic tanks, and non-anechoic tanks is calculated and analyzed. The results show that when the frequency is lower than the lowest normal frequency, the radiation impedance of the elastic spherical shell in anechoic tanks and non-anechoic tanks is consistent. When there are normal modes in non-anechoic tank, the radiation impedance of the elastic spherical shell varies with the normal frequencies. The proposed method can be used to calculate and analyze the radiation impedance of complex structures in different environments.

11:30

5aUWb9. Method for the measurement of the underwater transient sound characteristics *in a reverberation tank*. Xinyue Yu, Rui Tang, Qi Li, and Junming Zhang (Harbin Eng. Univ., Nantong Str., Nangang Dist No.145, Harbin 150001, China, tangrui@hrbeu.edu.cn)

A method to evaluate the sound power of transient sound sources in reverberation tanks is proposed and tested. The method is based on the steady state sound field characteristics in closed space, which are analyzed by normal-wave theory. To eliminate the interference caused by the boundary of the sound field, the volume integral of the spatial mean-square pressure in local space is measured and the sound power density of the non-anechoic is then obtained. Using the relationship between the sound field in the enclosed space and that of the free field, the sound power of the sound source can be obtained using the measured spatial mean-square pressure. The sound power of an impulsive sound generated by a spherical sound source is measured in a reverberation tank and the results are compared with the sound power measured in anechoic pool to verify the accuracy of the proposed method. The proposed method can be used to measure the radiated sound power of different types of transient sound sources in reverberation tanks.

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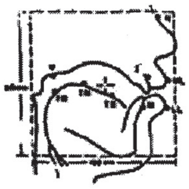

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Applicants may apply for one of four grades of membership, depending on their qualifications: Student Member, Associate Member, Corresponding Electronic Associate Member or full Member. To apply for Student Membership, fill out Parts I and II of this form; to apply for Associate, Corresponding Electronic Associate, or full Membership, or to transfer to these grades, fill out Parts I and III.

PART I. TO BE COMPLETED BY ALL APPLICANTS (Please print or type all entries)

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Part I Continued ➔

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.)

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PART II: APPLICATION FOR STUDENT MEMBERSHIP

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DEGREE EXPECTED	MONTH & YEAR DEGREE EXPECTED	NUMBER OF SEMESTER HOURS ATTENDED THIS SEMESTER
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SIGNATURES OF THE TWO FACULTY MEMBERS LISTED ABOVE CERTIFYING THAT YOU ARE REGISTERED AT LEAST HALF TIME		
SIGNATURE OF APPLICANT		DATE

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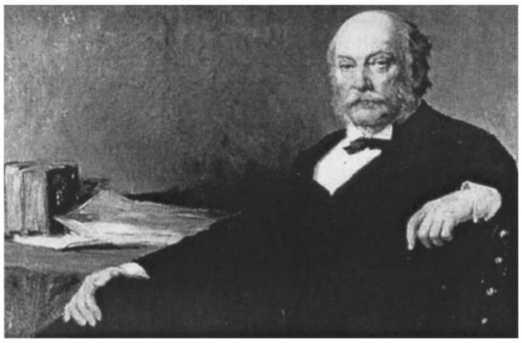

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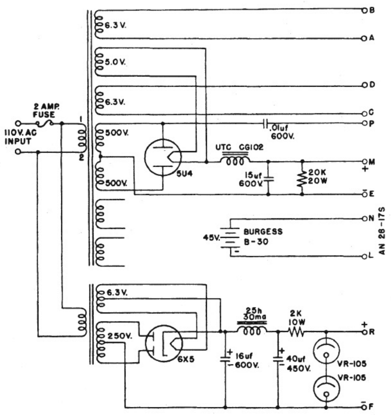

The Harvard Acoustics Research Laboratory was established in 1946 to support basic research in acoustics. Research results were disseminated formally by means of reports called technical memoranda (TMs). This CD includes the 61 reports issued between 1946 and 1971, when the contract with the Office of Naval Research was completed.

About half the TMs are doctoral theses in report form though some incorporate substantial additions. Most of the other half represent output by the postdoctoral fellows. The collection is introduced by David T. Blackstock of the University of Texas at Austin and brief bios for all of the TM authors are included.

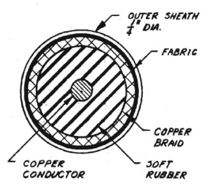

Fig.5. Schematic section showing cable makeup.

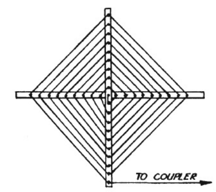

Fig.6. Configuration of the spiral mounting for the cable.

Having an unusually broad range for a single research group, the topics represented by the TMs fall mainly in the following categories: radiation, propagation, and scattering; bubbles and cavitation; acoustical instruments; electroacoustic transducers; and properties of solids, liquids, and gases.

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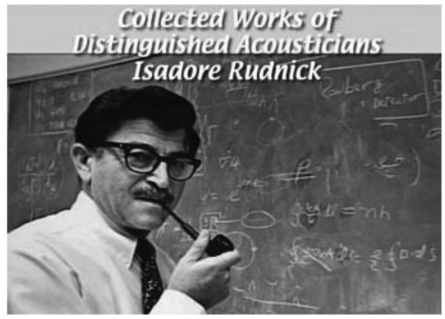
Collected Works of Distinguished Acousticians

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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