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# TECHNICAL PROGRAM SUMMARY
## Acoustics Virtually Everywhere
## 7—11 December 2011
***Indicates Special Session**


**KEYNOTE LECTURE**

*1pID Keynote Lecture: A Digital Stethoscope with Active Noise Suppression and Automatic Detection of Abnormalities in Lung Sounds

**ACOUSTICAL OCEANOGRAPHY**

1aAOa General Topics in Acoustical Oceanography I
1aAOb General Topics in Acoustical Oceanography II
1pAOa General Topics in Acoustical Oceanography III
1pAOb General Topics in Acoustical Oceanography IV
*3pAO Munk Award Lecture
4aAOa General Topics in Acoustical Oceanography V
4aAOb General Topics in Acoustical Oceanography VI
4pAO General Topics in Acoustical Oceanography VII

**ANIMAL BIOACOUSTICS**

1aAB Machine Learning for Classifying Animal Bioacoustic Signals
1pAB Innovative Tools for Animal Bioacoustics
*2aABa Celebrating Peter Narins' Contributions to Auditory Science I
*2aABb Celebrating Peter Narins' Contributions to Auditory Science II
*2pABa Celebrating Peter Narins' Contributions to Auditory Science III
*2pABb Celebrating Peter Narins' Contributions to Auditory Science IV
3aAB Animal Bioacoustics Poster Session
3pAB Parameters and Features of Animal Bioacoustic Signals
4aAB Effects of Anthropogenic Noise on Animal Behavior
4pAB Anthropogenic Sounds and Animal Bioacoustics
5aAB Passive Acoustic Monitoring of Animal Bioacoustic Signals

**ARCHITECTURAL ACOUSTICS**

*1aAAa Acoustics in Healthcare: Guidelines, Human Response, and the Way Forward Part I
*1aAAb Acoustics in Healthcare: Guidelines, Human Response, and the Way Forward Part II
*1pAAa Acoustics in Healthcare: Guidelines, Human Response, and the Way Forward Part III
1pAAb Architectural Acoustics Potpourri I
*2aAAa Sound Transmission and Impact Noise in Buildings I
*2aAAb Sound Transmission and Impact Noise in Buildings II
*2pAAa Sound Transmission and Impact Noise in Buildings III
*2pAAb Sound Transmission and Impact Noise in Buildings IV
*3aAAa Session in Memory of Jiri Tichy I
*3aAAb Session in Memory of Jiri Tichy II
*3pAAa New Developments in Classroom Acoustics I
*3pAAb Session in Memory of Jiri Tichy III
*3pAAc New Developments in Classroom Acoustics II
*4aAAa Session in Honor of William J. Cavanaugh I
*4aAAb Session in Honor of William J. Cavanaugh II
*4pAAa Session in Honor of William J. Cavanaugh III
*4pAAb Session in Honor of William J. Cavanaugh IV
5aAAa Architectural Acoustics Potpourri II
5aAAb Architectural Acoustics Potpourri III
5pAAa Architectural Acoustics Potpourri IV
5pAAb Architectural Acoustics Potpourri V

**BIOMEDICAL ACOUSTICS**

*1aBAa Death to Delay and Sum: Advanced Beamforming I
1aBAb General Biomedical Acoustics: Elastography I
*1aBAc Death to Delay and Sum: Advanced Beamforming II
1aBAd General Biomedical Acoustics: Elastography II
*1pBAa Death to Delay and Sum: Advanced Beamforming III
1pBAb General Biomedical Acoustics: Imaging I
1pBAc General Biomedical Acoustics: Imaging II
*2aBAa Modeling and Measuring Nonlinear Ultrasound Signals I
2aBAb General Biomedical Acoustics: Therapeutics & Elastography I
*2aBAc Modeling and Measuring Nonlinear Ultrasound Signals II
2aBAd General Biomedical Acoustics: Therapeutics & Elastography II
*2pBAa Modeling and Measuring Nonlinear Ultrasound Signals III
2pBAb General Biomedical Acoustics: Backscatter I
*2pBAc Modeling and Measuring Nonlinear Ultrasound Signals IV
2pBAd General Biomedical Acoustics: Backscatter II
*3aBAa Fractional Calculus Models of Compressional and Shear Waves for Medical Ultrasound I
3aBAb General Biomedical Acoustics: Tissue Engineering
*3aBAc Fractional Calculus Models of Compressional and Shear Waves for Medical Ultrasound II
3aBAd General Biomedical Acoustics: Quantitative Ultrasound
*3pBAa Fractional Calculus Models of Compressional and Shear Waves for Medical Ultrasound III
3pBAb General Biomedical Acoustics: Transcranial Focused Ultrasound
*4aBAa New Developments in Lung Ultrasound I
*4aBAb New Developments in Lung Ultrasound II
*4pBAa New Developments in Lung Ultrasound III
*4pBAb New Developments in Lung Ultrasound IV
5aBAa General Biomedical Acoustics: Therapeutics I
5aBAb General Biomedical Acoustics: Therapeutics II
5pBAa General Biomedical Acoustics: Therapeutics III
5pBAb General Biomedical Acoustics: Therapeutics IV

**COMPUTATIONAL ACOUSTICS**

*1aCAa Domain Truncation Techniques for Exterior Problems I
*1aCAb Domain Truncation Techniques for Exterior Problems II
*2aCAa Ray Methods Across Acoustics I
*2aCAb Ray Methods Across Acoustics II
*2pCA Ray Methods Across Acoustics III
*3aCAa Acoustic Optimization: Methods and Applications I
*3aCAb Acoustic Optimization: Methods and Applications II
*3pCA Acoustic Optimization: Methods and Applications III
4aCAa General Topics in Computational Acoustics I
4aCAb General Topics in Computational Acoustics II
4pCAa General Topics in Computational Acoustics III
4pCAb General Topics in Computational Acoustics IV

**EDUCATION IN ACOUSTICS**

*2aED Acoustics Education Prize Lecture
*3aEDa Acoustics Demonstrations for Classroom Teaching I
*3aEDb Acoustics Demonstrations for Classroom Teaching II
*3pED Hands-On Demonstrations: Acoustics at Home
4aEDa General Topics in Acoustics Education
*4aEDb Undergraduate Research Symposium Poster Session

**ENGINEERING ACOUSTICS**

1aEA General Topics in Engineering Acoustics I
1pEA General Topics in Engineering Acoustics II
*3aEAa Microphones: From Rock Stars to Rockets I
*3aEAb Microphones: From Rock Stars to Rockets II
*3pEAa Microphones: From Rock Stars to Rockets III
*3pEAb Microphones: From Rock Stars to Rockets IV
*4aEA Advanced Materials for Acoustic Transducers
4pEA General Topics in Engineering Acoustics III
5aEA General Topics in Engineering Acoustics IV

**INTERDISCIPLINARY**

*3pID Hot Topics in Acoustics
*4pIDa Graduate Programs in Acoustics Poster Session I
*4pIDb Graduate Programs in Acoustics Poster Session II
*4pIDc Acoustics in the COVID-19 Pandemic

**MUSICAL ACOUSTICS**

*2aMU Musical Acoustics Education at the Undergraduate Level I
*2pMUa Musical Acoustics Education at the Undergraduate Level II
*2pMUb Musical Acoustics Education at the Undergraduate Level III
3aMU General Topics in Musical Acoustics I
3pMUa General Topics in Musical Acoustics II
3pMUb General Topics in Musical Acoustics III
4aMUa General Topics in Musical Acoustics IV

4aMUb General Topics in Musical Acoustics V
4pMU General Topics in Musical Acoustics VI (Poster Session)

**NOISE**

1aNSa General Topics in Noise I
1aNSb General Topics in Noise II
*1pNSa Larry H. Royster Memorial Session I
*1pNSb Larry H. Royster Memorial Session II
*2aNSa Forty-One Years of Responding to External Stimuli: A Session in Honor of Elliott Berger I
*2aNSb Forty-One Years of Responding to External Stimuli: A Session in Honor of Elliott Berger II
*2pNSa Forty-One Years of Responding to External Stimuli: A Session in Honor of Elliott Berger III
*2pNSb Perception of Vehicle Noise I
*2pNSc Impact of Transportation Noise on Buildings
*3aNSa In Memory of Richard Lyon I
*3aNSb In Memory of Richard Lyon II
*3pNSa In Memory of Richard Lyon III
*3pNSb Perception of Vehicle Noise II
4aNSa Jet and Rocket Noise I
*4aNSb Noise Standards - Applications, Measurement Methods, Analyses, and Data Processing Involving ANSI/ASA S12 and/or ISO Noise Standards I
4aNSc Jet and Rocket Noise II
*4aNSd Noise Standards - Applications, Measurement Methods, Analyses, and Data Processing Involving ANSI/ASA S12 and/or ISO Noise Standards II
*4pNSa Noise Standards - Applications, Measurement Methods, Analyses, and Data Processing Involving ANSI/ASA S12 and/or ISO Noise Standards III
4pNSb General Topics in Noise III
4pNSc General Topics in Noise IV
*4pNSd Noise Standards - Applications, Measurement Methods, Analyses, and Data Processing Involving ANSI/ASA S12 and/or ISO Noise Standards III
*5aNSa Advances in Hearing Protection Devices I
*5aNSb Advances in Hearing Protection Devices II
*5pNSa Advances in Hearing Protection Devices III
*5pNSb Advances in Hearing Protection Devices IV

**PHYSICAL ACOUSTICS**

1aPAa General Topics: Acoustic Characterization of Materials I
1aPAb General Topics: Elastic Wave Propagation I
1aPAc General Topics: Elastic Wave Propagation II
1aPAd General Topics: Ultrasonics
1pPAa General Topics: Infrasound
1pPAb General Topics: Potpourri I
1pPAc General Topics: Bubbles I
*2aPAa Acoustical Measurements Through Optical Principles I
*2aPAb Acoustical Measurements Through Optical Principles II
*2pPAa Acoustical Measurements Through Optical Principles III
*2pPAb Acoustical Measurements Through Optical Principles IV
2pPAc General Topics: Bubbles II
3aPAa General Topics: Potpourri II
3aPAb General Topics: Turbulent Media
3aPAc General Topics: Jet Noise and Aeroacoustic Measurements
3pPAa General Topics: Acoustic Characterization of Materials II
3pPAb General Topics: Acoustic Characterization of Materials III
*4aPAa Acoustofluidics I
*4aPAb Acoustofluidics II
*4pPAa Acoustofluidics III
*4pPAb Acoustofluidics IV
*5aPAa Acoustofluidics V
*5aPAb Acoustofluidics VI
5pPAa General Topics: Potpourri III
5pPAb General Topics: Potpourri IV

**PSYCHOLOGICAL AND PHYSIOLOGICAL ACOUSTICS**

1aPPa Spectro-Temporal Processing and Music Perception (Poster Session)
1aPPb Speech Perception (Poster Session)
1aPPc Physiology (Poster Session)
*2aPPa Honoring William Yost's Contributions to Psychological Acoustics I
*2aPPb Honoring William Yost's Contributions to Psychological Acoustics II
*2pPPa Honoring William Yost's Contributions to Psychological Acoustics III
*2pPPb Honoring William Yost's Contributions to Psychological Acoustics IV
3aPP Binaural Hearing (Poster Session)
3pPP Localization and Unmasking
4aPPa Clinical and Aging Populations (Poster Session)
4aPPb Remote Testing for Auditory and Speech Research I (Poster Session)
4aPPc Remote Testing for Auditory and Speech Research II (Poster Session)
4pPP Remote Testing for Auditory and Speech Research III
5aPPa New Approaches and Virtual Reality I
5aPPb New Approaches and Virtual Reality II

**SIGNAL PROCESSING IN ACOUSTICS**

*1aSPa Random Matrix Theory in Acoustics I
*1aSPb Random Matrix Theory in Acoustics II
1pSPa General Topics in Signal Processing in Acoustics I
1pSPb General Topics in Signal Processing in Acoustics II
*2aSPa Acoustic Localization 1
*2aSPb Knowledge Discovery and Information Representation for Signal Processing in Acoustics I
*2aSPc Acoustic Localization II
*2aSPd Knowledge Discovery and Information Representation for Signal Processing in Acoustics II
*2pSPa Acoustic Localization III
*2pSPb Knowledge Discovery and Information Representation for Signal Processing in Acoustics III
*2pSPc Acoustic Localization IV
3aSPa General Topics in Signal Processing in Acoustics III
*3aSPb Acoustic Localization V
*3pSPa Machine Learning in Acoustics I
*3pSPb Machine Learning in Acoustics II
*4aSPa Machine Learning in Acoustics III
*4aSPb Machine Learning in Acoustics IV
*4pSPa Machine Learning in Acoustics V
*4pSPb Machine Learning in Acoustics VI
5aSPa General Topics in Signal Processing in Acoustics IV
5aSPb General Topics in Signal Processing in Acoustics V

**SPEECH COMMUNICATION**

1aSCa Clinical Studies in Speech I (Poster Session)
1aSCb Speech Production I (Poster Session)
1aSCc Speech Production II (Poster Session)
1pSCa Memory and Learning in Speech (Poster Session)
1pSCb Language Acquisition and Development (Poster Session)
1pSCc Speech Perception (Poster Session)
1pSCd Neurolinguistics and Psycholinguistics (Poster Session)
2aSCa Reintroducing the High-Frequency Region to Speech Perception Research I
2aSCb Reintroducing the High-Frequency Region to Speech Perception Research II
*2pSCa Reintroducing the High-Frequency Region to Speech Perception Research III
2pSCb Speech Articulation I (Poster Session)
2pSCc Clinical Topics in Speech II (Poster Session)
*3aSCa Listening in Challenging Circumstances I
*3aSCb Listening in Challenging Circumstances II
*3pSCa Listening in Challenging Circumstances III (Poster Session)
*3pSCb Listening in Challenging Circumstances IV (Poster Session)
3pSCc Speech Articulation II (Poster Session)
3pSCd Speech Production in Second Language I (Poster Session)
3pSCe Speech Production in Second Language II (Poster Session)
4aSCa Listening in Challenging Circumstances V (Poster Session)
4aSCb Prosody I (Poster Session)
4aSCc Prosody II (Poster Session)
4pSCa Phonetics of Race and Gender (Poster Session)
4pSCb Accent Perception and Talker Evaluation (Poster Session)

| | |
|---|---|
| 4pSCc | Audiovisual Speech (Poster Session) |
| *5aSCa | Developing a Cross-Platform Federated Code Repository for Speech Research I |
| *5aSCb | Developing a Cross-Platform Federated Code Repository for Speech Research II |
| 5pSCa | Speech Corpora and Modeling (Poster Session) |
| 5pSCb | Speech Production and Perception (Poster Session) |
| 5pSCc | Speech Perception in Second Language (Poster Session) |

**STRUCTURAL ACOUSTICS AND VIBRATION**

| | |
|---|---|
| *1pSAa | Acoustic Metamaterials I |
| *1pSAb | Acoustic Metamaterials II |
| *2aSAa | Acoustic Metamaterials III |
| *2aSAb | Acoustic Metamaterials IV |
| *2pSAa | Acoustic Metamaterials V |
| *2pSAb | Acoustic Metamaterials VI |
| *3aSAa | Non-Contact Vibration Measurement Methods I |
| *3aSAb | Non-Contact Vibration Measurement Methods II |
| *4aSAa | Active or Tunable Structural Acoustics I |
| *4aSAb | Active or Tunable Structural Acoustics II |
| 5aSAa | General Topics in Structural Acoustics I |
| 5aSAb | General Topics in Structural Acoustics II |

**UNDERWATER ACOUSTICS**

| | |
|---|---|
| 2aUWa | Array Processing in the Ocean |
| 2aUWb | Seabed Acoustics |
| 2pUWa | Detection and Localization in the Ocean I |
| 2pUWb | Detection and Localization in the Ocean II |
| 3aUW | Sources of Underwater Sound |
| 4aUWa | Underwater Acoustic Inversions |
| 4aUWb | Scattering in the Ocean |
| 4pUW | Underwater Acoustic Signal Processing |
| 5aUWa | Underwater Sound Transmission I |
| 5aUWb | Underwater Sound Transmission II |

## TECHNICAL PROGRAM CALENDAR
## Acoustics Virtually Everywhere
## 7–11 December 2020
## All times US Eastern Standard Time (EST)

**Monday Morning**

9:30 1aAAb **Architectural Acoustics, ASA Committee on Standards, Noise, and Speech Communication:** Acoustics in Healthcare: Guidelines, Human Response, and the Way Forward Part I

11:15 1aAAb **Architectural Acoustics, ASA Committee on Standards, Noise, and Speech Communication:** Acoustics in Healthcare: Guidelines, Human Response, and the Way Forward Part II

9:30 1aAB **Animal Bioacoustics:** Machine Learning for Classifying Animal Bioacoustic Signals

9:30 1aAOa **Acoustical Oceanography:** General Topics in Acoustical Oceanography I

11:15 1aAOb **Acoustical Oceanography:** General Topics in Acoustical Oceanography II

9:30 1aBAa **Biomedical Acoustics and Signal Processing in Acoustics:** Death to Delay and Sum:
Advanced Beamforming I

9:30 1aBAb **Biomedical Acoustics:** General Biomedical Acoustics: Elastography I

11:15 1aBAc **Biomedical Acoustics and Signal Processing in Acoustics:** Death to Delay and Sum: Advanced Beamforming II

11:15 1aBAd **Biomedical Acoustics:** General Biomedical Acoustics: Elastography II

9:30 1aCAa **Computational Acoustics and Structural Acoustics and Vibration:** Domain Truncation Techniques for Exterior Problems I

11:15 1aCAb **Computational Acoustics and Structural Acoustics and Vibration:** Domain Truncation Techniques for Exterior Problems II

9:30 1aEA **Engineering Acoustics:** General Topics in Engineering Acoustics I

9:30 1aNSa **Noise:** General Topics in Noise I

11:15 1aNSb **Noise:** General Topics in Noise II

9:30 1aPAa **Physical Acoustics:** General Topics: Acoustic Characterization of Materials I

9:30 1aPAb **Physical Acoustics:** General Topics: Elastic Wave Propagation I

11:15 1aPAc **Physical Acoustics:** General Topics: Elastic Wave Propagation II

11:15 1aPAd **Physical Acoustics:** General Topics: Ultrasonics

9:30 1aPPa **Psychological and Physiological Acoustics:** Spectro-Temporal Processing and Music Perception (Poster Session)

10:15 1aPPb **Psychological and Physiological Acoustics:** Speech Perception (Poster Session)

11:15 1aPPc **Psychological and Physiological Acoustics:** Physiology (Poster Session)

9:30 1aSCa **Speech Communication:** Clinical Studies in Speech I (Poster Session)

10:15 1aSCb **Speech Communication:** Speech Production I (Poster Session)

11:15 1aSCc **Speech Communication:** Speech Production II (Poster Session)

9:30 1aSPa **Signal Processing in Acoustics and Underwater Acoustics:** Random Matrix Theory in Acoustics I

11:15 1aSPb **Signal Processing in Acoustics and Underwater Acoustics:** Random Matrix Theory in Acoustics II

**Monday Afternoon**

1:05 1pAAa **Architectural Acoustics, ASA Committee on Standards, Noise, and Speech Communication:** Acoustics in Healthcare: Guidelines, Human Response, and the Way Forward Part III

2:50 1pAAb **Architectural Acoustics:** Architectural Acoustics Potpourri I

1:05 1pAB **Animal Bioacoustics:** Innovative Tools for Animal Bioacoustics

1:05 1pAOa **Acoustical Oceanography:** General Topics in Acoustical Oceanography III

2:50 1pAOb **Acoustical Oceanography:** General Topics in Acoustical Oceanography IV

1:05 1pBAa **Biomedical Acoustics and Signal Processing in Acoustics:** Death to Delay and Sum: Advanced Beamforming III

1:05 1pBAb **Biomedical Acoustics:** General Biomedical Acoustics: Imaging I

2:50 1pBAc **Biomedical Acoustics:** General Biomedical Acoustics: Imaging II

2:50 1pEA **Engineering Acoustics:** General Topics in Engineering Acoustics II

4:30 1pID **Interdisciplinary:** Keynote Lecture: A Digital Stethoscope with Active Noise Suppression and Automatic Detection of Abnormalities in Lung Sounds

1:05 1pNSa **Noise and Education in Acoustics:** Larry H. Royster Memorial Session I

2:50 1pNSb **Noise and Education in Acoustics:** Larry H. Royster Memorial Session II

1:05 1pPAa **Physical Acoustics:** General Topics: Infrasound

1:05 1pPAb **Physical Acoustics:** General Topics: Potpourri I

2:50 1pPAc **Physical Acoustics:** General Topics: Bubbles I

1:05 1pSAa **Structural Acoustics and Vibration, Physical Acoustics, and Engineering Acoustics:** Acoustic Metamaterials I

2:50 1pSAb **Structural Acoustics and Vibration, Physical Acoustics, and Engineering Acoustics:** Acoustic Metamaterials II

12:00 1pSCa **Speech Communication:** Memory and Learning in Speech (Poster Session)

1:50 1pSCb **Speech Communication:** Language Acquisition and Development (Poster Session)

2:50 1pSCc **Speech Communication:** Speech Perception (Poster Session)

3:35 1pSCd **Speech Communication:** Neurolinguistics and Psycholinguistics (Poster Session)

1:05 1pSPa **Signal Processing in Acoustics:** General Topics in Signal Processing in Acoustics I

2:50 1pSPb **Signal Processing in Acoustics:** General Topics in Signal Processing in Acoustics II

**Tuesday Morning**

9:30 2aAAa **Architectural Acoustics, Noise, ASA Committee on Standards, and Structural Acoustics and Vibration:** Sound Transmission and Impact Noise in Buildings I

11:15 2aAAb **Architectural Acoustics, Noise, ASA Committee on Standards, and Structural Acoustics and Vibration:** Sound Transmission and Impact Noise in Buildings II

9:30 2aABa **Animal Bioacoustics and Psychological and Physiological Acoustics:** Celebrating Peter Narins' Contributions to Auditory Science I

11:15 2aABb **Animal Bioacoustics and Psychological and Physiological Acoustics:** Celebrating Peter Narins' Contributions to Auditory Science II

9:30 2aBAa **Biomedical Acoustics, Physical Acoustics, Signal Processing in Acoustics, and Computational Acoustics:** Modelling and Measuring Nonlinear Ultrasound Signals I

9:30 2aBAb **Biomedical Acoustics:** General Biomedical Acoustics: Therapeutics & Elastography I

11:15 2aBAc **Biomedical Acoustics, Physical Acoustics, Signal Processing in Acoustics, and Computational Acoustics:** Modelling and Measuring Nonlinear Ultrasound Signals II

11:15 2aBAd **Biomedical Acoustics:** General Biomedical Acoustics: Therapeutics & Elastography II

9:30 2aCAa **Computational Acoustics, Underwater Acoustics, Physical Acoustics, and Structrual Acoustics and Vibration:** Ray Methods Across Acoustics I

11:15 2aCAb **Computational Acoustics, Underwater Acoustics, Physical Acoustics, and Structrual Acoustics and Vibration:** Ray Methods Across Acoustics II

11:15 2aED **Education in Acoustics:** Acoustics Education Prize Lecture

11:15 2aMU **Musical Acoustics and Education in Acoustics:** Musical Acoustics Education at the Undergraduate Level I

9:30 2aNSa **Noise and ASA Committee on Standards:** Forty-One Years of Responding to External Stimuli: A Session in Honor of Elliott Berger I

11:15 2aNSb **Noise and ASA Committee on Standards:** Forty-One Years of Responding to External Stimuli: A Session in Honor of Elliott Berger II

9:30 2aPAa **Physical Acoustics, Biomedical Acoustics, and Musical Acoustics:** Acoustical Measurements Through Optical Principles I

11:15 2aPAb **Physical Acoustics, Biomedical Acoustics, and Musical Acoustics:** Acoustical Measurements Through Optical Principles II

9:30 2aPPa **Psychological and Physiological Acoustics:** Honoring William Yost's Contributions to Psychological Acoustics I

11:15 2aPPb **Psychological and Physiological Acoustics:** Honoring William Yost's Contributions to Psychological Acoustics II

9:30 2aSAa **Structural Acoustics and Vibration, Physical Acoustics, and Engineering Acoustics:** Acoustic Metamaterials III

11:15 2aSAb **Structural Acoustics and Vibration, Physical Acoustics, and Engineering Acoustics:** Acoustic Metamaterials IV

9:30 2aSCa **Speech Communication and Psychological and Physiological Acoustics:** Reintroducing the High-Frequency Region to Speech Perception Research I

11:15 2aSCb **Speech Communication and Psychological and Physiological Acoustics:** Reintroducing the High-Frequency Region to Speech Perception Research II

9:30 2aSPa **Signal Processing in Acoustics, Acoustical Oceanography, Animal Bioacoustics, Engineering Acoustics, Underwater Acoustics, Noise,and Architectural Acoustics:** Acoustic Localization I

9:30 2aSPb **Signal Processing in Acoustics, Animal Bioacoustics, Underwater Acoustics, Computational Acoustics, and Acoustical Oceanography:** Knowledge Discovery and Information Representation for Signal Processing in Acoustics I

11:15 2aSPc **Signal Processing in Acoustics, Acoustical Oceanography, Animal Bioacoustics, Engineering Acoustics, Underwater Acoustics, Noise,and Architectural Acoustics:** Acoustic Localization II

11:15 2aSPd **Signal Processing in Acoustics, Animal Bioacoustics, Underwater Acoustics, Computational Acoustics, and Acoustical Oceanography:** Knowledge Discovery and Information Representation for Signal Processing in Acoustics II

9:30 2aUWa **Underwater Acoustics:** Array Processing in the Ocean

11:15 2aUWb **Underwater Acoustics:** Seabed Acoustics

**Tuesday Afternoon**

1:05 2pAAa **Architectural Acoustics, Noise, ASA Committee on Standards, and Structural Acoustics and Vibration:** Sound Transmission and Impact Noise in Buildings III

2:50 2pAAb **Architectural Acoustics, Noise, ASA Committee on Standards, and Structural Acoustics and Vibration:** Sound Transmission and Impact Noise in Buildings IV

1:05 2pABa **Animal Bioacoustics and Psychological and Physiological Acoustics:** Celebrating Peter Narins' Contributions to Auditory Science III

2:50 2pABb **Animal Bioacoustics and Psychological and Physiological Acoustics:** Celebrating Peter Narins' Contributions to Auditory Science IV

1:05 2pBAa **Biomedical Acoustics, Physical Acoustics, Signal Processing in Acoustics, and Computational Acoustics:** Modelling and Measuring Nonlinear Ultrasound Signals III

1:05 2pBAb **Biomedical Acoustics:** General Biomedical Acoustics: Backscatter I

2:50 2pBAc **Biomedical Acoustics, Physical Acoustics, Signal Processing in Acoustics, and Computational Acoustics:** Modelling and Measuring Nonlinear Ultrasound Signals IV

2:50 2pBAd **Biomedical Acoustics:** General Biomedical Acoustics: Backscatter II

1:05 2pCA **Computational Acoustics, Underwater Acoustics, Physical Acoustics, and Structrual Acoustics and Vibration:** Ray Methods Across Acoustics III

1:05 2pMUa **Musical Acoustics and Education in Acoustics:** Musical Acoustics Education at the Undergraduate Level II

2:50 2pMUb **Musical Acoustics and Education in Acoustics:** Musical Acoustics Education at the Undergraduate Level III

1:05 2pNSa **Noise and ASA Committee on Standards:** Forty-One Years of Responding to External Stimuli: A Session in Honor of Elliott Berger III

2:50 2pNSb **Noise, Structural Acoustics and Vibration, ASA Committee on Standards, Psychological and Physiological Acoustics:** Perception of Vehicle Noise I

2:50 2pNSc **Noise, Architectural Acoustics, Structural Acoustics and Vibration, and ASA Committee on Standards:** Impact of Transportation Noise on Buildings

1:05 2pPAa **Physical Acoustics, Biomedical Acoustics, and Musical Acoustics:** Acoustical Measurements Through Optical Principles III

2:50 2pPAb **Physical Acoustics, Biomedical Acoustics, and Musical Acoustics:** Acoustical Measurements Through Optical Principles IV

2:50 2pPAc **Physical Acoustics:** General Topics: Bubbles II

1:05 2pPPa **Psychological and Physiological Acoustics:** Honoring William Yost's Contributions to Psychological Acoustics III

2:50 2pPPb **Psychological and Physiological Acoustics:** Honoring William Yost's Contributions to Psychological Acoustics IV

1:05 2pSAa **Structural Acoustics and Vibration, Physical Acoustics, and Engineering Acoustics:** Acoustic Metamaterials V

2:50 2pSAb **Structural Acoustics and Vibration, Physical Acoustics, and Engineering Acoustics:** Acoustic Metamaterials VI

1:05 2pSCa **Speech Communication and Psychological and Physiological Acoustics:** Reintroducing the High-Frequency Region to Speech Perception Research III

2:50 2pSCb **Speech Communication:** Speech Articulation I (Poster Session)

3:35 2pSCc **Speech Communication:** Clinical Topics in Speech II (Poster Session)

1:05 2pSPa **Signal Processing in Acoustics, Acoustical Oceanography, Animal Bioacoustics, Engineering Acoustics, Underwater Acoustics, Noise, and Architectural Acoustics:** Acoustic Localization III

1:05 2pSPb **Signal Processing in Acoustics, Animal Bioacoustics, Underwater Acoustics, Computational Acoustics, and Acoustical Oceanography:** Knowledge Discovery and Information Representation for Signal Processing in Acoustics III

2:50 2pSPc **Signal Processing in Acoustics, Acoustical Oceanography, Animal Bioacoustics, Engineering Acoustics, Underwater Acoustics, Noise, and Architectural Acoustics:** Acoustic Localization IV

1:05 2pUWa **Underwater Acoustics:** Detection and localization in the Ocean I

2:50 2pUWb **Underwater Acoustics:** Detection and Localization in the Ocean II

**Wednesday Morning**

9:30 3aAAa **Architectural Acoustics, Engineering Acoustics, Noise, and Signal Processing in Acoustics:** Session in Memory of Jiri Tichy I

11:15 3aAAb **Architectural Acoustics, Engineering Acoustics, Noise, and Signal Processing in Acoustics:** Session in Memory of Jiri Tichy II

9:30 3aAB **Animal Bioacoustics:** Animal Bioacoustics Poster Session

9:30 3aBAa **Biomedical Acoustics, Physical Acoustics, Structural Acoustics and Vibration, and \Computational Acoustics:** Fractional Calculus Models of Compressional and Shear Waves for Medical Ultrasound I

9:30 3aBAb **Biomedical Acoustics:** General Biomedical Acoustics: Tissue Engineering

11:15 3aBAc **Biomedical Acoustics, Physical Acoustics, Structural Acoustics and Vibration, and Computational Acoustics:** Fractional Calculus Models of Compressional and Shear Waves f for Medical Ultrasound II

11:15 3aBAd **Biomedical Acoustics:** General Biomedical Acoustics: Quantitative Ultrasound

9:30 3aCAa **Computational Acoustics, Structural Acoustics and Vibration, Underwater Acoustics, and Acoustical Oceanography:** Acoustic Optimization: Methods and Applications I

11:15 3aCAb **Computational Acoustics, Structural Acoustics and Vibration, Underwater Acoustics, and Acoustical Oceanography:** Acoustic Optimization: Methods and Applications II

9:30 3aEAa **Engineering Acoustics, Architectural Acoustics, and ASA Committee on Standards:** Microphones: From Rock Stars to Rockets I

11:15 3aEAb **Engineering Acoustics, Architectural Acoustics, and ASA Committee on Standards:** Microphones: From Rock Stars to Rockets II

9:30 3aEDa **Education in Acoustics, Musical Acoustics, Physical Acoustics, Structural Acoustics and Vibration, Noise, and Architectural Acoustics:** Acoustics Demonstrations for Classroom Teaching I

11:15 3aEDb **Education in Acoustics, Musical Acoustics, Physical Acoustics, Structural Acoustics and Vibration, Noise, and Architectural Acoustics:** Acoustics Demonstrations for Classroom Teaching II

11:15 3aMU **Musical Acoustics:** General Topics in Musical Acoustics I

9:30 3aNSa **Noise, Structural Acoustics and Vibration, and Architectural Acoustics:** In Memory of Richard Lyon I

11:15 3aNSb **Noise, Structural Acoustics and Vibration, and Architectural Acoustics:** In Memory of Richard Lyon II

9:30 3aPAa **Physical Acoustics:** General Topics: Potpourri II

9:30 3aPAb **Physical Acoustics:** General Topics: Turbulent Media

11:15 3aPAc **Physical Acoustics:** General Topics: Jet Noise and Aeroacoustic Measurements

9:30 3aPP **Psychological and Physiological Acoustics:** Binaural Hearing (Poster Session)

9:30 3aSAa **Structural Acoustics and Vibration, Engineering Acoustics, Physical Acoustics, and Musical Acoustics:** Non-Contact Vibration Measurement Methods I

11:15 3aSAb **Structural Acoustics and Vibration, Engineering Acoustics, Physical Acoustics, and Musical Acoustics:** Non-Contact Vibration Measurement Methods II

9:30 3aSCa **Speech Communication, Noise, Architectural Acoustics, and Psychological and Physiological Acoustics:** Listening in Challenging Circumstances I

11:15 3aSCb **Speech Communication, Noise, Architectural Acoustics, and Psychological and Physiological Acoustics:** Listening in Challenging Circumstances II

9:30 3aSPa **Signal Processing in Acoustics:** General Topics in Signal Processing in Acoustics III

11:15 3aSPb **Signal Processing in Acoustics, Acoustical Oceanography, Animal Bioacoustics, Engineering Acoustics, Underwater Acoustics, Noise, and Architectural Acoustics:** Acoustic Localization V

9:30 3aUW **Underwater Acoustics:** Sources of Underwater Sound

**Wednesday Afternoon**

1:05 3pAAa **Architectural Acoustics, ASA Committee on Standards, Noise, Education in Acoustics,and Speech Communication:** New Developments in Classroom Acoustics I

1:05 3pAAb **Architectural Acoustics, Engineering Acoustics, Noise, and Signal Processing in Acoustics:** Session in Memory of Jiri Tichy III

2:50 3pAAc **Architectural Acoustics, ASA Committee on Standards, Noise, Education in Acoustics, and Speech Communication:** New Developments in Classroom Acoustics II

1:05 3pAB **Animal Bioacoustics:** Parameters and Features of Animal Bioacoustic Signals

2:50 3pAO **Acoustical Oceanography:** Munk Award Lecture

1:05 3pBAa **Biomedical Acoustics, Physical Acoustics, Structural Acoustics and Vibration, and Computational Acoustics:** Fractional Calculus Models of Compressional and Shear Waves for Medical Ultrasound III

1:05 3pBAb **Biomedical Acoustics:** General Biomedical Acoustics: Transcranial Focused Ultrasound

1:05 3pCA **Computational Acoustics, Structural Acoustics and Vibration, Underwater Acoustics and Acoustical Oceanography:** Acoustic Optimization: Methods and Applications III

1:05 3pEAa **Engineering Acoustics, Architectural Acoustics, and ASA Committee on Standards:** Microphones: From Rock Stars to Rockets III

2:50 3pEAb **Engineering Acoustics, Architectural Acoustics, and ASA Committee on Standards:** Microphones: From Rock Stars to Rockets IV

1:05 3pED **Education in Acoustics and Women in Acoustics:** Hands-On Demonstrations: Acoustics at Home

2:50 3pID **Interdisciplinary:** Hot Topics in Acoustics

1:05 3pMUa **Musical Acoustics:** General Topics in Musical Acoustics II

2:50 3pMUb **Musical Acoustics:** General Topics in Musical Acoustics III

1:05 3pNSa **Noise, Structural Acoustics and Vibration, and Architectural Acoustics:** In Memory of Richard Lyon III

2:50 3pNSb **Noise, Structural Acoustics and Vibration, ASA Committee on Standards, Psychological and Physiological Acoustics:** Perception of Vehicle Noise II

1:05 3pPAa **Physical Acoustics:** General Topics: Acoustic Characterization of Materials II

2:50 3pPAb **Physical Acoustics:** General Topics: Acoustic Characterization of Materials III

1:05 3pPP **Psychological and Physiological Acoustics:** Localization and Unmasking

12:00 3pSCa **Speech Communication, Noise, Architectural Acoustics, and Psychological and Physiological Acoustics:** Listening in Challenging Circumstances III (Poster Session)

1:05 3pSCb **Speech Communication, Noise, Architectural Acoustics, and Psychological and Physiological Acoustics:** Listening in Challenging Circumstances IV (Poster Session)

1:50 3pSCc **Speech Communication:** Speech Articulation II (Poster Session)

2:50 3pSCd **Speech Communication:** Speech Production in Second Language I (Poster Session)

3:35 3pSCe **Speech Communication:** Speech Production in Second Language II (Poster Session)

1:05 3pSPa **Signal Processing in Acoustics, Animal Bioacoustics, Underwater Acoustics, Computational Acoustics and Acoustical**

**Oceanography:** Machine Learning in Acoustics I

2:50 3pSPb **Signal Processing in Acoustics, Animal Bioacoustics, Underwater Acoustics, Computational Acoustics, and Acoustical Oceanography:** Machine Learning in Acoustics II

**Thursday Morning**

9:30 4aAAa **Architectural Acoustics:** Session in Honor of William J. Cavanaugh I

11:15 4aAAb **Architectural Acoustics:** Session in Honor of William J. Cavanaugh II

9:30 4aAB **Animal Bioacoustics:** Effects of Anthropogenic Noise on Animal Behavior

9:30 4aAOa **Acoustical Oceanography:** General Topics in Acoustical Oceanography V

11:15 4aAOb **Acoustical Oceanography:** General Topics in Acoustical Oceanography VI

9:30 4aBAa **Biomedical Acoustics and Signal Processing in Acoustics:** New Developments in Lung Ultrasound I

11:15 4aBAb **Biomedical Acoustics and Signal Processing in Acoustics:** New Developments in Lung Ultrasound II

9:30 4aCAa **Computational Acoustics:** General Topics in Computational Acoustics I

11:15 4aCAb **Computational Acoustics:** General Topics in Computational Acoustics II

9:30 4aEA **Engineering Acoustics:** Advanced Materials for Acoustic Transducers

9:30 4aEDa **Education in Acoustics:** General Topics in Acoustics Education

11:15 4aEDb **Education in Acoustics:** Undergraduate Research Symposium Poster Session

9:30 4aMUa **Musical Acoustics:** General Topics in Musical Acoustics IV

11:15 4aMUb **Musical Acoustics:** General Topics in Musical Acoustics V

9:30 4aNSa **Noise:** Jet and Rocket Noise I

9:30 4aNSb **Noise, ASA Committee on Standards, Architectural Acoustics, and Signal Processing in Acoustics:** Noise Standards - Applications, Measurement Methods, Analyses, and Data Processing Involving ANSI/ASA S12 and/or ISO Noise Standards I

11:15 4aNSc **Noise:** Jet & Rocket Noise II

11:15 4aNSd **Noise, ASA Committee on Standards, Architectural Acoustics, and Signal Processing in Acoustics:** Noise Standards - Applications, Measurement Methods, Analyses, and Data Processing Involving ANSI/ASA S12 and/or ISO Noise Standards II

9:30 4aPAa **Physical Acoustics, Biomedical Acoustics, Engineering Acoustics, and Structural Acoustics and Vibration:** Acoustofluidics I

11:15 4aPAb **Physical Acoustics, Biomedical Acoustics, Engineering Acoustics, and Structural Acoustics and Vibration:** Acoustofluidics II

9:30 4aPPa **Psychological and Physiological Acoustics:** Clinical and Aging Populations(Poster Session)

10:15 4aPPb **Psychological and Physiological Acoustics:** Remote Testing for Auditory and Speech Research I (Poster Session)

11:15 4aPPc **Psychological and Physiological Acoustics:** Remote Testing for Auditory and Speech Research II (Poster Session)

9:30 4aSAa **Structural Acoustics and Vibration, Engineering Acoustics, Signal Processing in Acoustics, and Noise:** Active or Tunable Structural Acoustics I

11:15 4aSAb **Structural Acoustics and Vibration, Engineering Acoustics, Signal Processing in Acoustics, and Noise:** Active or Tunable Structural Acoustics II

9:30 4aSCa **Speech Communication, Noise, Architectural Acoustics, and Psychological and Physiological Acoustics:** Listening in Challenging Circumstances V (Poster Session)

10:15 4aSCb **Speech Communication:** Prosody I (Poster Session)

11:15 4aSCc **Speech Communication:** Prosody II (Poster Session)

9:30 4aSPa **Signal Processing in Acoustics, Animal Bioacoustics, Underwater Acoustics, Computational Acoustics, and Acoustical Oceanography:** Machine Learning in Acoustics III

1:15 4aSPb **Signal Processing in Acoustics, Animal Bioacoustics, Underwater Acoustics, Computational Acoustics, and Acoustical Oceanography:** Machine Learning in Acoustics IV

9:30 4aUWa **Underwater Acoustics:** Underwater Acoustic Inversions

11:15 4aUWb **Underwater Acoustics:** Scattering in the Ocean

**Thursday Afternoon**

1:05 4pAAa **Architectural Acoustics:** Session in Honor of William J. Cavanaugh III

2:50 4pAAb **Architectural Acoustics:** Session in Honor of William J. Cavanaugh IV

1:05 4pAB **Animal Bioacoustics:** Anthropogenic Sounds and Animal Bioacoustics

1:05 4pAO **Acoustical Oceanography:** General Topics in Acoustical Oceanography VII

1:05 4pBAa **Biomedical Acoustics and Signal Processing in Acoustics:** New Developments in Lung Ultrasound III

2:50 4pBAb **Biomedical Acoustics and Signal Processing in Acoustics:** New Developments in Lung Ultrasound IV

1:05 4pCAa **Computational Acoustics:** General Topics in Computational Acoustics III

2:50 4pCAb **Computational Acoustics:** General Topics in Computational Acoustics IV

1:05 4pEA **Engineering Acoustics:** General Topics in Engineering Acoustics III

1:05 4pIDa **Interdisciplinary:** Graduate Programs in Acoustics Poster Session I

1:50 4pIDb **Interdisciplinary:** Graduate Programs in Acoustics Poster Session II

3:35 4pIDc **Interdisciplinary:** Acoustics in the COVID-19 Pandemic

1:05 4pMU **Musical Acoustics:** General Topics in Musical Acoustics VI (Poster Session)

1:05 4pNSa **Noise, ASA Committee on Standards, Architectural Acoustics, and Signal Processing in Acoustics:** Noise Standards - Applications, Measurement Methods, Analyses, and Data Processing Involving ANSI/ASA S12 and/or ISO Noise Standards III

1:05 4pNSb **Noise:** General Topics in Noise III

2:50 4pNSc **Noise:** General Topics in Noise IV

2:50 4pNSd **Noise, ASA Committee on Standards, Architectural Acoustics, and Signal Processing in Acoustics:** Noise Standards - Applications, Measurement Methods, Analyses, and Data Processing Involving ANSI/ASA S12 and/or ISO Noise Standards IV

1:05 4pPAa **Physical Acoustics, Biomedical Acoustics, Engineering Acoustics, and Structural Acoustics and Vibration:** Acoustofluidics III

2:50 4pPAb **Physical Acoustics, Biomedical Acoustics, Engineering Acoustics, and Structural Acoustics and Vibration:** Acoustofluidics IV

12:00 4pPP **Psychological and Physiological Acoustics:** Remote Testing for Auditory and Speech Research III

1:05 4pSCa **Speech Communication:** Phonetics of Race and Gender (Poster Session)

1:50 4pSCb **Speech Communication:** Accent Perception and Talker Evaluation (Poster Session)

2:50 4pSCc **Speech Communication:** Audiovisual Speech (Poster Session)

1:05 4pSPa **Signal Processing in Acoustics, Animal Bioacoustics, Underwater Acoustics, Computational Acoustics, and Acoustical Oceanography:** Machine Learning in Acoustics V

2:50 4pSPb **Signal Processing in Acoustics, Animal Bioacoustics, Underwater Acoustics, Computational Acoustics, and Acoustical Oceanography:** Machine Learning in Acoustics VI

2:50 4pUW **Underwater Acoustics:** Underwater Acoustic Signal Processing

**Friday Morning**

9:30 5aAAa **Architectural Acoustics:** Architectural Acoustics Potpourri II

11:15 5aAAb **Architectural Acoustics:** Architectural Acoustics Potpourri III

9:30 5aAB **Animal Bioacoustics:** Passive Acoustic Monitoring of Animal Bioacoustic Signals

9:30 5aBAa **Biomedical Acoustics:** General Biomedical Acoustics: Therapeutics I

11:15 5aBAb **Biomedical Acoustics:** General Biomedical Acoustics: Therapeutics II

9:30 5aEA **Engineering Acoustics:** General Topics in Engineering Acoustics IV

9:30 5aNSa **Noise, Psychological and Physiological Acoustics, and ASA Committee on Standards:** Advances in Hearing Protection Devices I

11:15 5aNSb **Noise, Psychological and Physiological Acoustics, and ASA Committee on Standards:** Advances in Hearing Protection Devices II

9:30 5aPAa **Physical Acoustics, Biomedical Acoustics, Engineering Acoustics, and Structural Acoustics and Vibration:** Acoustofluidics V

11:15 5aPAb **Physical Acoustics, Biomedical Acoustics, Engineering Acoustics, and Structural Acoustics and Vibration:** Acoustofluidics VI

9:30 5aPPa **Psychological and Physiological Acoustics:** New Approaches and Virtual Reality I

11:15 5aPPb **Psychological and Physiological Acoustics:** New Approaches and Virtual Reality II

9:30 5aSAa **Structural Acoustics and Vibration:** General Topics in Structural Acoustics I

11:15 5aSAb **Structural Acoustics and Vibration:** General Topics in Structural Acoustics II

9:30 5aSCa **Speech Communication:** Developing a Cross-Platform Federated Code Repository for Speech Research I

11:15 5aSCb **Speech Communication:** Developing a Cross-Platform Federated Code Repository for Speech Research II

9:30 5aSPa **Signal Processing in Acoustics:** General Topics in Signal Processing in Acoustics IV

11:15 5aSPb **Signal Processing in Acoustics:** General Topics in Signal Processing in Acoustics V

9:30 5aUWa **Underwater Acoustics:** Underwater Sound transmission I

11:15 5aUWb **Underwater Acoustics:** Underwater Sound transmission II

**Friday Afternoon**

1:05 5pAAa **Architectural Acoustics:** Architectural Acoustics Potpourri IV

2:50 5pAAb **Architectural Acoustics:** Architectural Acoustics Potpourri V

1:05 5pBAa **Biomedical Acoustics:** General Biomedical Acoustics: Therapeutics III

2:50 5pBAb **Biomedical Acoustics:** General Biomedical Acoustics: Therapeutics IV

1:05 5pNSa **Noise, Psychological and Physiological Acoustics, and ASA Committee on Standards:** Advances in Hearing Protection Devices III

2:50 5pNSb **Noise, Psychological and Physiological Acoustics, and ASA Committee on Standards:** Advances in Hearing Protection Devices IV

1:05 5pPAa **Physical Acoustics:** General Topics: Potpourri III

2:50 5pPAb **Physical Acoustics:** General Topics: Potpourri IV

1:05 5pSCa **Speech Communication:** Speech Corpora and Modeling (Poster Session)

1:50 5pSCb **Speech Communication:** Speech Production and Perception (Poster Session)

2:50 5pSCc **Speech Communication:** Speech Perception in Second Language (Poster Session)

# SCHEDULE OF COMMITTEEE MEETINGS AND OTHER EVENTS

**Times are Eastern Standard Time (EST)**

## TECHNICAL COMMITTEEE OPEN MEETINGS

| | |
|---|---|
| Tue, 8 December, 4:30 p.m. | Acoustical Oceanography |
| Tue, 8 December, 4:30 p.m. | Animal Bioacoustics |
| Tue, 8 December, 4:30 p.m. | Architectural Acoustics |
| Tue, 8 December, 4:30 p.m. | Engineering Acoustics |
| Tue, 8 December, 4:30 p.m. | Physical Acoustics |
| Tue, 8 December, 4:30 p.m. | Psychological and Physiological Acoustics |
| Tue, 8 December, 6:30 p.m. | Signal Processing in Acoustics |
| Tue, 8 December, 4:30 p.m. | Structural Acoustics and Vibration |
| Thu, 10 December, 4:30 p.m. | Biomedical Acoustics |
| Thu, 10 December, 4:30 p.m. | Computational Acoustics |
| Thu, 10 December, 4:30 p.m. | Musical Acoustics |
| Thu, 10 December, 6:00 p.m. | Noise |
| Thu, 10 December, 4:30 p.m. | Speech Communication |
| Thu, 5 December, 4:30 p.m. | Underwater Acoustics |

## MEEETING SERVICES, SPECIAL EVENTS, SOCIAL EVENTS

| | |
|---|---|
| Mon-Fri, 7-11 December | Registration |
| 8:30 a.m. - 4:00 p.m. | |
| Mon-Fri, 7-11 December | Committee Information Booths |
| 9:30 a.m. - 4:00 p.m. | |
| Mon, 7 December | Keynote Lecture |
| 4:30 p.m. to 6:00 p.m. | |
| Mon, 7 December | New Student Orientation |
| 6:00 p.m. - 6:30 p.m. | |
| Mon, 7 December | Student Meet and Greet |
| 6:30 p.m. - 7:30 p.m. | |
| Mon, 7 December | Student Social |
| 7:30 p.m. - 8:30 p.m. | |
| Wed, 9 December | Effective Negotiation and Difficult Conversations |
| 2:50 p.m. to 4:20 p.m. | Skills and Strategies |
| Plenary Session/Awards Ceremony | |
| 4:30 p.m. - 6:00 p.m. | |
| Wed. 9 December | Social Hour |
| 6:30 p.m. - 8:00 p.m. | |
| Wed, 9 December, | ASA Jam |
| 8:00 p.m. - midnight. . . | |

# Acoustics Virtually Everywhere,
# 179th Meeting of the Acoustical Society of America

Acoustics Virtually Everywhere, the 179th meeting of the Acoustical Society of America will be held Monday through Friday, 7-11 December 2020.

**SECTION HEADINGS**

1. REGISTRATION
2. ACCESSIBILITY
3. KEYNOTE LECTURE
4. TECHNICAL SESSIONS
5 TECHNICAL SESSION DESIGNATIONS
6. HOT TOPICS SESSION
7. THE WALTER MUNK AWARD AND LECTURE
8. ROSSING PRIZE IN ACOUSTICS EDUCATION AND THE EDUCATION IN ACOUSTICS PRIZE LECTURE
9. TECHNICAL COMMITTEE OPEN MEETINGS
10. INFORMATION BOOTHS
11. ANNUAL MEETING OF ASA MEMBERS
12. PLENARY SESSION AND AWARDS CEREMONY
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18. STUDENT EVENTS: NEW STUDENT ORIENTATION, MEET AND GREET, FELLOWSHIP AND GRANT PANEL, STUDENT RECEPTION
19. JAM SESSION
20. TECHNICAL PROGRAM ORGANIZING COMMITTEE
21. PHOTOGRAPHING AND RECORDING
22. ABSTRACT ERRATA

**1. REGISTRATION**

Registration is required for all attendees.

Registration will open Monday through Friday, 7-11 December, from 9:00 a.m. to 4:00 p.m.

The registration fees (in USD) are $399 for members of the Acoustical Society of America; $550 for non-members, $125 for ASA Student members, $225 for students who are not members of ASA.

One-day registration is available at $250 for members and $325 for nonmembers (one-day means attending the meeting on only one day either to present a paper and/or to attend sessions).

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 $225.

There is a two-step procedure to register. First visit https://asa_ave20.vfairs.com/en/registration and create an account on the meeting platform. Then proceed to the payment site at https://www.associationsciences.org/ASAM/.

**2. 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.

**3. KEYNOTE LECTURE**

The Keynote Lecture titled A digital stethoscope with active noise suppression and automatic detection of abnormalities in lung sounds will be presented by James E. West and Ellington West on Monday, 7 December, 4:30 p.m. to 6:00 p.m.

James West, an ASA Past President and Gold Medal recipient, is a Professor at Johns Hopkins University. Ellington West is Co-Founder and CEO of Sonavi Labs, creator of medical devices and software rooted in AI and applied to auscultation, the act of listening to body sounds.

**4. TECHNICAL SESSIONS**

The technical program includes 1120 abstracts organized into 213 lecture sessions and 31 poster sessions for presentation during the meeting.

Lecture sessions are a maximum of 90-minutes long and contain up to 4 15-minute recorded presentations. Each recording will be followed by 3 minutes for live Q&A with the author and 2 minutes for attendees to move to other sessions. If an author has withdrawn his/her 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.

The poster gallery includes over 330 posters organized into 31 45 minute sessions containing10 posters each. Poster presenters will be available for live text chats during the scheduled session and may also arrange additional chat times.

**5. TECHNICAL SESSION DESIGNATIONS**

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

1-Monday, 7 December
2-Tuesday, 8 December
3-Wednesday, 9 December
4-Thursday, 10 December
5-Friday, 11 December

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

AA Architectural Acoustics
AB Animal Bioacoustics
AO Acoustical Oceanography
BA Biomedical Acoustics
CA Computational Acoustics

EA Engineering Acoustics
ED Education in Acoustics
ID Interdisciplinary
MU Musical Acoustics
NS Noise
PA Physical Acoustics
PP Psychological and Physiological Acoustics
SA Structural Acoustics and Vibration
SC Speech Communication
SP Signal Processing in Acoustics
UW Underwater Acoustics

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

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

### 6. HOT TOPICS SESSION

The Hot Topics session (3pID) will be held on Wednesday, 9 December, at 2:50 p.m. Papers will be presented on current topics in the fields of Architectural Acoustics, Musical Acoustics, and Psychological and Physiological Acoustics.

### 7. THE WALTER MUNK AWARD AND LECTURE

The 2020 Walter Munk Award will be presented to Larry Mayer, University of New Hampshire, by the Oceanography Society and Dr. Mayer will present the Munk Award Lecture

*Can We Map the Entire Global Ocean Seafloor by 2030?* on Wednesday, 9 December, in session 3pAO at 2:55 p.m.

### 8. ROSSING PRIZE IN ACOUSTICS EDUCATION AND ACOUSTICS EDUCATION PRIZE LECTURE

The 2020 Rossing Prize in Acoustics Education will be presented to Daniel Butko, University of Oklahoma, at the Plenary Session on Wednesday, 9 December. Dr. Butko will present the Acoustics Education Prize Lecture titled 2aED1: *Over a decade of decibels – celebrating teaching architectural acoustics within an architecture curriculum to students with various majors and minors* on Tuesday, 8 December, in session 2aED at t 11:20 a.m.

### 9. TECHNICAL COMMITTEE OPEN MEETINGS

Technical Committees will hold open meetings on Tuesday and Thursday. The schedule and rooms for each Committee meeting are given on 2592.

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.

### 10. INFORMATION BOOTHS

Several ASA committees and organizations that publish ASA content will be hosting information booths in the Exhibit Hall. Information about the committees’ activities will be available for download and the booths may also include video and other content of interest to attendees.

### 11. ANNUAL MEETING OF MEMBERS

The annual meeting of ASA Members, which is required by New York State law, will be held online on Wednesday, 9 December 2020, at approximately 4:30 p.m., at Acoustics Virtually Everywhere, the 179th Meeting of the Acoustical Society of America.

### 12. PLENARY SESSION AND AWARDS CEREMONY

A plenary session will be held Wednesday, 9 December, at 4:30 p.m.

The session will begin with the ASA Annual Meeting of Members. ASA scholarship recipients and the David T. Blackstock Student Council Mentor Award recipient will be introduced. Award presentations will include the Rossing Prize in Acoustics Education, R. Bruce Lindsay Award, Silver Medal in Noise, Wallace Clement Sabine Medal, and the Gold Medal. Fellows elected at the San Diego and spring 2020 virtual meetings and add will be announced. See 2664 for a list of fellows and award recipients.

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

### 13. ANSI STANDARDS COMMITTEES

Meetings of ANSI Accredited Standards Committees will not be held at Acoustics Virtually Everywhere.

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

### 14. TECHNICAL SESSION BREAKS

There will be 15-minute breaks each morning after the first group of technical sessions are presented. There will also be a 20-minute break between morning and afternoon sessions each day.

### 15. GATHER.TOWN AND CHAT LOUNGES

The ASA Virtual Technology Task Force has arranged for the use of Gather.Town for social networking. There are also Chat Rooms that are available on the meeting platform which provide for text chats.

Gather.town is a video chat platform that has avatars move around a map. As you get close to other avatars, your video’s will pop up and you will be able to chat. By moving your avatar around using the arrow keys on your keyboard, you can have spontaneous conversations with those around you.

These can be either one-on-one or small groups depending on how many people are around you and what you set your interaction distance to be.

Links to Gather.town and the Chat Lounges will be provided at the meeting site.

### 16. PROCEEDINGS OF MEETINGS ON ACOUSTICS (POMA)

The Acoustics Virtually Everywhere 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.

### 17. SOCIAL HOUR

The ASA Society Social is scheduled for Wednesday, 9 December, immediately following the Plenary session at approximately 6:30 p.m. Join other conference attendees for a session of fun virtual games and for the chance to win prizes! Test your acoustics knowledge. Cheer on your favorite teams. Connect with fellow attendees. This will be a fun social hour before the amazing ASA Jam.

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

Follow the student twitter throughout the meeting @ ASAStudents.

The ASA Student Council is hosting three social events during Acoustics Virtually Everywhere. All three of these events will be held on Monday, 7 December, via Zoom.

The first event is New Student Orientation, which will take place 6:00 p.m.—6:30 p.m. (EST). The next event, the Student Meet and Greet will be held 6:30 p.m.--7:30 p.m. This event will include Acoustics Trivia in breakout rooms with prizes going to the teams who score the highest. Following trivia, there will be time for student social discussion via their teams' breakout rooms.

The final student event is the Student Social, scheduled for 7:30 p.m.--8:30 p.m. (EST). The Student Social will include 1-minute student presentations as a part of the "Poster Slam" as well as an "Introduce Your Pet" event (a separate sign-up email will be sent for these events). Finally, the student social will provide time for students to introduce themselves and get to know one another via breakout rooms.

### 19. JAM SESSION

You are invited to the ASA JAM on Wednesday night, 4 December, from 8:00 p.m. midnight. . .

The ASA Jam is a popular music social event held at ASA meetings. For Acoustics Virtually Everywhere, the ASA Jam Subcommittee has opened up two online platforms for meeting attendees to share recorded music that they have written and/or performed in an effort to facilitate virtual interaction and sharing.

### 20. TECHNICAL PROGRAM ORGANIZING COMMITTEE

Brandon Cudequest, Technical Program Chair; David Knobles, Acoustical Oceanography; Benjamin Taft, Animal Bioacoustics; David Manley, Benjamin Bridgewater, Architectural Acoustics; Kang Kim, Demi Libertario, Biomedical Acoustics; Jennifer Cooper, Computational Acoustics; Michael Haberman, Caleb Sieck, Engineering Acoustics; Daniel Russell, Education in Acoustics; Peter Rucz, Musical Acoustics; William Murphy, James Phillips, Hales Swift; Noise; Kevin Lee, Sam Wallen, Physical Acoustics; Ellen Peng, Psychological and Physiological Acoustics; Erick Dieckman, David Geroski, Signal Processing in Acoustics; Yoonjeong Lee, Susannah Lee, Matthew Faytak, Rajka Smiljanic, Speech Communication; Benjamin Shafer, Anthony Bonomo, Robert M. Koch, Structural Acoustics and Vibration; Derek Olson, Underwater Acoustics; Kieren Smith, Student Council.

### 21. PHOTOGRAPHING AND RECORDING

Photographing and recording of videos and posters are not permitted without prior permission from the Acoustical Society.

### 22. ABSTRACT ERRATA

This meeting program is Part 2 of the October 2020 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*.

# FIFTY-YEAR AWARDS

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

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# TWENTY-FIVE YEAR AWARDS

"Silver" certificates will be sent to the following individuals have been members of the Society for twenty-five years:

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## Session 1aAAa

## Architectural Acoustics, Noise, and Speech Communication: Acoustics in Healthcare: Guidelines, Human Response, and the Way Forward Part I

Jay M. Bliefnick, Cochair
*Acentech, 33 Moulton St., Cambridge, MA 02138*

Kenneth W. Good, Cochair
*Armstrong World Industries, Inc., 2500 Columbia Avenue, Lancaster, PA 17601*

**Chair's Introduction—9:30**

### *Invited Papers*

**9:35**

**1aAAa1. A guide to acoustics healthcare guidelines.** Jay Bliefnick (Acentech, 33 Moulton St., Cambridge, MA 02138, jbliefnick@gmail.com)

Over the past few decades a number of guidelines describing recommended acoustics performance have been developed for hospitals and healthcare environments. These include documents from the Federal Guidelines Institute (FGI), World Health Organization (WHO), ANSI/ASA, ASHRAE, and LEED. Each of these guidelines include numerous acoustical recommendations (some that even contradict one another), regarding such topics as background noise levels in patient rooms, sound isolation requirements, reverberation times, and speech privacy. These guidelines strive to improve the overall acoustical comfort within healthcare environments in addition to patient perception, as measured in surveys like the Hospital Consumer Assessment of Healthcare Providers and Systems (HCAHPS). In this project, the implementation of hospital guidelines in the United States was assessed, focusing on when and where specific guidelines were instituted, the impact these guidelines have had on acoustics performance, and what can be done to improve acoustics conditions within hospitals and healthcare environments across the country.

**9:55**

**1aAAa2. Flooring impact sound: An overview of the new test standard, its application for healthcare design, and potential updates to the standard.** Michael Raley (Ecore Int., 715 Fountain Ave., Lancaster, PA 17601, mike.raley@ecoreintl.com)

Noise is a major concern in many healthcare environments, and the "Quietness of Hospital Environment" item on the Hospital Consumer Assessment of Healthcare Providers and Systems (HCAHPS) consistently receives one of the lowest scores. A recent study showed that among many acoustic metrics, peak occurrence rates were one of the best correlated with HCAHPS scores for Quietness of Hospital Environment. Flooring impact sound from footsteps, rolling carts, and dropped items can be a significant source of noise in hospitals, and that impact sound is likely to contribute to peak occurrence rates. Flooring-generated noise is often a concern for healthcare designers, and is a primary consideration when picking flooring materials, but until recently there has been no standard for assessing the floor impact sound radiation of various floor coverings. A new standard, ASTM E3133, provides this assessment. In this presentation, data from recent testing to the ASTM E3133 standard will be given for various floor coverings. Additionally, the presentation will cover issues with the test methodology and potential solutions. Finally, preliminary results from a proposed field test standard will be discussed.

**10:15**

**1aAAa3. The ASA/ANSI S12.60 criteria for evaluating speech privacy in healthcare and the way forward.** Kenneth W. Good (Armstrong World Industries, Inc., 2500 Columbia Ave., Lancaster, PA 17601, kwgoodjr@armstrong.com)

Speech privacy in healthcare spaces is more than just a good business practice. Although speech privacy is required within the HIPAA privacy rule, it is more than just a legal requirement. Speech privacy in healthcare spaces can be a life safety issue. Without confidence of speech privacy, patients maybe more likely withhold personal and embarrassing information necessary for accurate diagnosis and treatment. This paper will review the current state of the ASA/ANSI S12.60 criteria for evaluating speech privacy in healthcare and discuss work moving forward.

10:35

**1aAAa4. Comprehensive multi-variable analysis of signal-to-noise ratio in dining and social spaces.** Noral D. Stewart (Stewart Acoust. Consultants, 7330 Chapel Hill Rd., Ste. 201, Raleigh, NC 27607, norals2020@sacnc.com) and Joseph F. Bridger (Stewart Acoust. Consultants, Raleigh, NC)

Difficult communication in dining and social areas is a common problem and the most common acoustical complaint in senior living and healthcare facilities. The signal to noise ratio is the major factor in this and is influenced strongly by architectural design. Prior simplified signal-to-noise-ratio analyses with few variables indicate impracticably high amounts of sound absorption required for acceptable results. This analysis adds the benefit of table-top reflections and the effects of the reverberant part of the signal (partially allotted to noise using known methods) and the direct part of the noise (using a closed-form approximation). It shows that the signal to noise ratio is strongly improved by table-top reflections and improved by (1) increased floor area per person (allowing more absorption and space between tables), (2) rooms with fewer people, (3) smaller table dimensions putting listeners closer to signal talkers, and (4) optimized discussion group size where possible to reduce the number of simultaneous talkers. Increased reverberation time due to higher ceilings is slightly detrimental. While a highly absorptive space is required, consideration of these other factors significantly reduces the required absorption to a more practical level.

MONDAY MORNING, 7 DECEMBER 2020 11:15 A.M. TO 12:45 P.M.

# Session 1aAAb

## Architectural Acoustics, Noise, and Speech Communication: Acoustics in Healthcare: Guidelines, Human Response, and the Way Forward Part II

Jay M. Bliefnick, Cochair
*Acentech, 33 Moulton St., Cambridge, MA 02138*

Kenneth W. Good, Cochair
*Armstrong World Industries, Inc., 2500 Columbia Avenue, Lancaster, PA 17601*

**Chair's Introduction—11:15**

### *Invited Paper*

11:20

**1aAAb1. The spatial uniformity of an electronic sound masking system in an open-plan space.** Joonhee Lee (Dept. of Bldg., Civil and Environ. Eng., Concordia Univ., Montreal, QC, Canada), Farideh Zarei (Dept. of Bldg., Civil and Environ. Eng., Concordia Univ., 1515 Rue Sainte-Catherine O., Montreal, QC H3G 2W1, Canada, f.zarei@softdb.com), Roderick Mackenzie, and Vincent Le Men (Soft dB, Monteal, QC, Canada)

An electronic sound-masking system reduces workers' distractions in open-plan spaces by utilizing an artificial broadband sound. The artificial sound should raise a background noise level spectrum to the targeted masking sound level uniformly over the entire area. Uneven distribution of the masking sound levels can cause unnecessary loud background noise or inefficient sound masking performance at the same time and in different locations. The ASTM E 1573-18 standard provides a procedure to quantify the uniformity of the masking sound but does not specify any acceptable degree of uniformity. Thus, this study aims to investigate the uniformity of the masking sound field in an open-plan space under varying room acoustic conditions. The acoustic measurement was carried out in an open-plan office with a measurement grid of 0.6 m. The spatial variation of the sound pressure levels was calculated with the measured one-third-octave band SPLs from 250 Hz to 4 kHz. The study also employed computer-aided acoustic simulation to find key design parameters, impacting the uniformity of the masking sound. The results show that the number of loudspeakers, a partition height, scattering, and absorption coefficients can significantly influence the spatial uniformity and speech privacy within the space. Finally, the results proposed an acceptable variation of the masking sound field by examining the Articulation Index (AI) change in the space.

**11:40**

**1aAAb2. Speech privacy in healthcare: A case study discussion of design and measurement results.** Nicole A. Kolak (Acoust., TEECOM, 1333 Broadway, Oakland, CA 94612, nicole.kolak@teecom.com) and Peter Holst (Acoust., TEECOM, Oakland, CA)

This presentation will be an overview of speech privacy design and measurement summary for a medical office building in Roseville, CA, where TEECOM provided the acoustical design services. Speech privacy can be analyzed in several ways. For this presentation, TEECOM will compare the measured speech privacy performance using Speech Privacy Predictor (SPP) and Speech Privacy Class (SPC) for partitions between exam rooms and between offices as well as their respective door-walls. The project encountered budget constraints that limited the design options; demising walls were required to be partial-height. An electronic sound masking system was incorporated to elevate the ambient noise levels within comfortable limits to provide a controlled background noise level in the hallways and in enclosed spaces. Post-occupancy comments on the quality of speech privacy were gathered by the Architecture client and will be shared to support the quantifiable findings. Considerations for further improvements are described, including space planning and other design elements that can be used to achieve the confidentially goals and requirements established by HIPAA (Health Insurance Portability and Accountability Act).

**12:00**

**1aAAb3. Exploratory analysis of noise reduction solutions in a psychiatric ward using beamforming techniques.** Mojtaba Navvab (Architecture, Univ. of Michigan, TCAUP, Arch Bldg., 2000 Bonisteel Blvd, Ann Arbor, MI 48109, moji@umich.edu)

The world health organization recommendations related to a psychiatric ward for sound pressure level measurements overtime are not to exceed 30 and 35 dBA, respectively. However, no published study results since 1975 have reported full compliance within a hospital. Hospitals are a stressful work environment regardless of the noise level that averages between 53 and 58 dBA. Unique working solutions offered by design architects and insurance companies to reduce noise conditions in a physiological health facility are not transferable to psychiatric units that are operating as a behavioral health facility. This study reports on an exploratory analysis of the noise reduction strategies within a psychiatric ward. Architecturally and electronically implemented noise mitigation solutions are used to minimize noise annoyance, which is related to the noise sensitivity of the patient and staff. Sound mapping and noise source localization are conducted utilizing beamforming techniques. Onsite noise measurements in time and frequency domain provide design opportunities for isolating patient rooms from noise by the public and staff working areas. The impact of working solutions on speech privacy and annoyance is determined. The results contribute to a more extensive database and opportunities toward improvement in design guidelines for psychiatric wards and a positive influence on the hospital soundscape.

**12:20**

**1aAAb4. Restorative effect of interactive virtual reality natural environment experience on psychophysiological responses.** Jin Yong Jeon (Dept. of Medical and Digital Eng., Hanyang Univ., 222 Wangsimni-ro, Seongdong-gu, Seoul 04763, South Korea, jyjeon@hanyang.ac.kr), Hyun In Jo, and Kounseok Lee (Dept. of Psychiatry, Hanyang Univ. Medical Ctr., Seoul, South Korea)

In this study, the restorative effect of a natural environment experience simulated with interactive virtual reality technology on the psychophysiological response was investigated. Three simulated virtual reality stimuli were selected: a green-dominated area, green and water combined area, and water-dominated area. Free movement was permitted to provide interaction in virtual reality, and an appropriate sound environment was provided in real time to each space through 3-D sound technology. Fifty subjects experiencing the normal levels of stress or depression participated in this study. The aim was to locate the most restorative point in the natural environment, and the physical properties at each point were quantified. In addition, to collect psychological responses, subjects were asked to respond to a restoration questionnaire based on the attention restoration and stress reduction theory and a questionnaire on soundscape and landscape perception. Concurrently, various physiological responses, such as heart rate, electroencephalogram, and eye tracking, were measured. Finally, the relationship between physical characteristics and psychophysiological responses was investigated. Based on this, guidelines were presented on how to create a virtual reality natural environment for stress recovery.

## Session 1aAB

# Animal Bioacoustics: Machine Learning for Classifying Animal Bioacoustic Signals

Ming Zhong
*Microsoft, 24927 SE 43 St., Issaquah, WA 98029*

**Chair's Introduction—9:30**

### *Contributed Papers*

**9:35**

**1aAB1. Bioacoustics and machine learning for automated avian species monitoring in global biodiversity hotspots.** Ming Zhong (Microsoft, 24927 SE 43rd St., Issaquah, WA 98029, mizhong@microsoft.com), Ruth Taylor, Damian Christey, Shane Palkovitz, Naomi Bates (Future Generations Univ., Franklin, WV), Rahul Dodhia, and Juan Lavista Ferres (Microsoft, Redmond, WA)

The complex realities of changing climate and biodiversity are often imperfectly understood. As a conservation tool, bioacoustic monitoring with machine learning (ML) models can provide valuable insights for informed decision making in conservation efforts. In this study, we built deep convolutional neural networks to classify calls of two rare bird species detected in ambient field recordings from the mountains of Nepal. With limited amount of training data, we used data augmentation techniques to effectively increase the size of training set and thus boost the model performance. The model output provides insights of species activity and abundance over time across multiple ecosystems, which can be used as a biodiversity change indicator, and also helps scientists and conservation experts to better understand species behavior, diversity, and habitat preference. This modeling methodology and its framework can be easily adopted by other acoustic classification problems.

**9:55**

**1aAB2. Temporal context improves automatic recognition of call sequences in soundscape data.** Shyam Madhusudhana (Ctr. for Conservation Bioacoustics, Cornell Lab of Ornithology, Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850, skm246@cornell.edu), Yu Shiu, Holger Klinck (Ctr. for Conservation Bioacoustics, Cornell Lab of Ornithology, Cornell Univ., Ithaca, NY), Erica Fleishman (College of Earth, Ocean, and Atmospheric Sci., Oregon State Univ., Corvallis, OR), Xiaobai Liu (Dept. of Comput. Sci., San Diego State Univ., San Diego, CA), Eva-Marie Nosal (Dept. of Ocean and Resources Eng., Univ. of Hawai'i at Má noa, Honolulu, HI), Tyler Helble (U.S. Navy, Space and Naval Warfare Systems Command, System Ctr. Pacific, San Diego, CA), Danielle Cholewiak (Northeast Fisheries Sci. Ctr., National Marine Fisheries Service, National Oceanic and Atmospheric Administration, Woods Hole, MA), Douglas Gillespie (Sea Mammal Res. Unit, Scottish Oceans Inst., Univ. of St. Andrews, St. Andrews, Scotland, United Kingdom), Ana Širović (Marine Biology Dept., Texas A&M Univ., Galveston, TX), and Marie A. Roch (Dept. of Comput. Sci., San Diego State Univ., San Diego, CA)

Convolutional neural networks (CNNs) are commonly employed for detecting animal vocalizations. We explored whether use of the temporal patterns in song notes can improve recognition. Fin whales (*Balaenoptera physalus*) produce sequences of low-frequency, down-swept calls (20 Hz pulses) over many minutes. Timing between calls can be exploited to improve detection. We trained a base CNN model to detect 20 Hz pulses in 4 s audio segments. Then, we trained three variants of long short-term memory (LSTM) networks to process sequences produced by the CNN. In the first, the inputs to the LSTM were the scalar prediction scores from the CNN. The second examined sequences of features produced by the CNN before classification. The third combined the feature vectors and scores produced by the CNN. We conducted cross-validation experiments on recordings from the Southern California Bight collected between 2008 and 2014. All three variants outperformed the CNN. The precision-recall (PR) curves of the hybrid models dominated that of the base model, with improvements of 8%–13% in both peak F1-score and area under PR-curve. The second and third hybrid variants performed better than the first. CNN-LSTM hybrid models efficiently improve recognition of call sequences by incorporating temporal context.

**10:15**

**1aAB3. Multispecies bioacoustics classification using transfer learning of deep convolutional neural networks with pseudo-labeling.** Ming Zhong (Microsoft, 24927 SE 43rd St., Issaquah, WA 98029, mizhong@microsoft.com), Jack LeBien, Marconi Campos-Cerqueira (Sieve Analytics, San Juan), Rahul Dodhia, Juan Lavista Ferres (Microsoft, Redmond, WA), Julian Velev, and T. Mitchell Aide (Sieve Analytics, San Juan)

In this study, we evaluated deep convolutional neural networks for classifying the calls of 24 birds and amphibian species detected in ambient field recordings from the tropical mountains of Puerto Rico. Training data were collected using a template-based detection algorithm and manually validated with a graphical interface. To reduce the labor intensive and time-consuming process of manual validation, as well as to increase the accuracy of species classification with acoustic recordings, we propose a novel approach that combines transfer learning of a pre-trained deep convolutional neural network (CNN) model, a semi-supervised pseudo-labeling method, and a custom training loss function. While generating sufficient training data is a major challenge for many deep learning applications, our proposed methodology enables the network to be trained in a supervised fashion with labeled and unlabeled data simultaneously, which effectively increases the size of training set and thus boosts the model performance. As a result, the model achieves 97.7% sensitivity, 96.4% specificity and 96.6% accuracy in classifying a test set of manually validated true and false positive template-based detections. This multi-label multi-species classification methodology and its framework can be easily expanded to other acoustic classification problems.

10:35

**1aAB4. Classification of marine mammal vocalizations using machine learning.** Vasilis Koutsomitopoulos (Appl. Res. Labs., Univ. of Texas at Austin, PO Box 8029, Austin, TX 78713-8029, taka.koutso@gmail.com) and Minh Le (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX)

Marine mammal vocalizations are a significant component of the underwater soundscape in many parts of the world. The ability to detect and classify these vocalizations has implications in marine mammal research and tracking, acoustic noise measurement and modeling, and surface and submerged ship navigation. Advances in machine learning and deep neural networks have enabled classification technologies that can match and even exceed human performance in many areas. In this talk, we will explore a supervised, multi-label classifier using a Convolutional Neural Network to discriminate different types of marine mammal vocalizations across species using spectrograms derived from single-channel audio recordings from the Watkins Marine Mammal Sound Database of the Woods Hole Oceanographic Institute. The Watkins database contains over seven decades of recordings of more than 60 professionally identified marine mammal species and is publicly available for scientific research. We will discuss in details the rationale for our data labeling and pre-processing, model training and evaluation, and data and model visualization in order to understand and improve classification performance and robustness.



MONDAY MORNING, 7 DECEMBER 2020 — 9:30 A.M. TO 11:00 A.M.

## Session 1aAOa

## Acoustical Oceanography: General Topics in Acoustical Oceanography I

Charles W. Holland, Chair

*Electrical and Computer Engineering, Portland State University, Portland, OR 97207*

**Chair's Introduction—9:30**

### *Contributed Papers*

9:35

**1aAOa1. Sediment sound speeds from a monostatic (multibeam) sonar.** Charles W. Holland (Elec. and Comput. Eng., Portland State Univ., Appl. Res. Lab., State College, PA 16804, charles.holland@pdx.edu) and Samuel Pinson (ENSTA, Brest, France)

Time-averaged sediment layer sound speeds, or interval velocities are generally obtained from source-receiver separations at various offsets. In marine environments, the traditional measurement technique employs a towed broadband source and a long towed receive array, often kilometers in length. A method is described for estimating interval velocities using a simpler, monostatic configuration. The method is first tested using simulated data from various layered seabed structures with roughness at each interface; the resulting estimated interval velocity is within less than 1% of the true value. Monostatic measured data from the Gulf of Lion are also presented which exhibit many characteristics similar to the simulated data. The method applied to the measured data yield an interval velocity of 1569 m/s in an 18 m sediment layer. This accords with nearby independent data from cores and wide-angle reflection analysis. [Work supported by ONR Littoral Geosciences and Optics program.]

9:55

**1aAOa2. A new measurement of the deepest depth of the ocean.** Scott Loranger (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 86 Water St., Woods Hole, MA 02543, sloranger@whoi.edu), David R. Barclay (Oceanogr., Dalhousie Univ., Halifax, NS, Canada), and Michael Buckingham (Marine Physical Lab., Scripps Inst. of Oceanogr., San Diego, CA)

Scientists and explorers have been searching to determine the exact location and depth of the deepest part of the ocean since the HMS Challenger made the first sounding of the Mariana Trench in 1875. The consensus is that the deepest abyss in the ocean is in the Challenger Deep, a portion of the Mariana Trench with depths greater than 10 000 m. Since the HMS Challenger II returned to the Mariana Trench in 1952, 14 estimates of the deepest depth of the ocean have been made. Estimates of the location and maximum depth are as diverse as the methods used including wire soundings, explosives, single and multibeam sonars, and remotely operated and manned submersibles. During an Office of Naval Research supported expedition to the Challenger Deep in 2014, two free falling passive acoustic instruments were deployed. The implosion of one instrument was recorded by the other when both were at depths greater than 8000 m. Multiple reflections from the seafloor and sea surface of the sound generated by the implosion were used to determine the depth of the Challenger Deep. The result was the most constrained estimate of the deepest part of the ocean, 10 991 $\pm$ 6 m.

10:15

**1aAOa3. A classification approach to the characterization of seabed geoacoustic profiles via deep learning.** David J. Forman (Phys., Hillsdale College, 60 E College St., Whitley 107A, Hillsdale, MI 49242, dforman@hillsdale.edu), Tracianne B. Neilsen, and David F. Van Komen (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Acoustic source localization algorithms in the shallow ocean require information about the ocean seabed. While past approaches have characterized seabed geoacoustic profiles via continuous physical parameters, a classification approach approximates the distribution of seabed types as a discrete set of seabed classes. To exclude acoustically indistinguishable seabeds from the discrete classes, the acoustic separation between two seabeds may be evaluated by the Pearson correlation between their characteristic transmission loss curves. This process yields a subset of discrete seabed classes. To distinguish these classes, we train a convolutional neural network classifier on the pressure time-series of SUS charges, which are simulated using the geoacoustic profiles of the seabed classes. This network successfully predicts seabed class, when tested on a holdout set where the sound speed profiles of the ocean are adequately similar to those in the training set. An acoustic separation measure between sound speed profiles, analogous to the one for seabeds, directly relates to the generalization error of the network. [Work supported by the Office of Naval Research Contract No. N00014-16-C-3065 and by the NSF REU program, Grant No. 1757998.]

10:35

**1aAOa4. Acoustic characterization of the New England shelf break.** Scott Loranger (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 86 Water St., Woods Hole, MA 02543, sloranger@whoi.edu) and Andone C. Lavery (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

The Southern New England shelf break is a dynamic region of the ocean; warm core rings shed by the Gulf stream, salt fingers, and intrusions all contribute to a complex spatiotemporally variable oceanographic environment. One of the most distinct features is the shelfbreak front where warm and salty deep water collides with lower density, colder and fresher shelf water. The meandering front coincides with the presence of a group of strong scatterers, believed to be fish with swim bladders, known to be present in the area. During the Office of Naval Research Sediment Characterization Experiment in 2017 acoustic backscatter from narrowband (18 and 38 kHz) and broadband (70–280 kHz) shipboard echo sounders detected the turbulent microstructure at the shelf break front. The sound speed, temperature, salinity, and density were inferred from the acoustic backscatter, and verified by CTD and expendable bathythermograph (XBT) casts. This method provides a more holistic view of the water masses at the shelfbreak front than is possible from CTD and XBT casts alone. The position of the strong scatterers in relation to the front was also characterized to elucidate the water masses and physical oceanographic processes that drive the distribution of the scatterers.

MONDAY MORNING, 7 DECEMBER 2020 11:15 A.M. TO 12:45 P.M.

## Session 1aAOb

## Acoustical Oceanography: General Topics in Acoustical Oceanography II

Miad Al Mursaline, Chair
*Mechanical Engineering/ Applied Ocean Physics & Engineering, Massachusetts Institute of Technology/Woods Hole Oceanographic Institution, 70 Pacific Street, Cambridge, MA 02139*

**Chair's Introduction—11:15**

### *Contributed Papers*

11:20

**1aAOb1. Acoustic scattering from cylinders for aquaculture applications.** Miad Al Mursaline (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 70 Pacific St., Cambridge, MA 02139, miad@mit.edu), Timothy K. Stanton, Andone C. Lavery, and Erin M. Fischell (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

In aquaculture farms, long-lines are used to grow kelp for food, pharmaceuticals, and biofuel. As these farms become larger and move offshore, acoustic sensing with high-frequency broadband echosounders is likely a scalable option for large-scale monitoring, making understanding of the scattering properties of the long-lines critical. However, the boundary conditions for the long-lines are complicated, and field data show that the long-lines occupy only a small number of Fresnel zones. To determine the influence of Fresnel zone interference, cylindrical targets with known boundary conditions but a small number of Fresnel zones insonified were studied. Using a formulation involving a monopole line source and an infinitely long cylinder excited at normal incidence relative to the cylinder axis, an apparent volume flow is derived for the case of a spherically spreading source with Bessel function transceiver directivity. Element pressure derived using this volume flow is integrated along the length of the cylinder to yield the scattered field. Theoretical results using this method are validated experimentally at angles near normal incidence. The theoretical formulation is then extended for larger angles by deriving the volume flow based on the analytic solution to an obliquely excited infinite elastic cylinder. [Work funded by ARPA-E.]

11:40

**1aAOb2. Echo-counting in the mesopelagic zone using a broadband split-beam acoustic scattering system.** Zhaozhong Zhuang (Woods Hole Oceanographic Inst., 98 Water St., Woods Hole, MA 02543, zzhuang@whoi.edu) and Andone C. Lavery (Woods Hole Oceanographic Inst., Woods Hole, MA)

The ocean mesopelagic zone (200–1000 m depths) is a largely unexplored and remote ecosystem. Recent studies have suggested that the biomass of mesopelagic fish may be up to ten times larger than previously reported, more that the rest of the ocean combined. Interpretation of shipboard measurements of mesopelagic scattering layers in terms of abundance and biomass is complicated by large sampling volumes, complex assemblages of organisms, and highly uncertain target strength models. To address some of these challenges, Deep-See, a towed sensor platform carrying multiple broadband acoustic systems, has been developed, allowing real-time measurements of target strength and abundance. The focus here is on a new broadband split-beam echosounder (5.5 kHz–18.5 kHz) deployed in summer 2018 and 2019 off the New England shelf. This system is used to estimate the density of mesopelagic organisms using echo-counting and echo-statistics approaches. The parameters selected for echo-counting are discussed in order to achieve accurate estimates of abundance and also to identify and track individual targets. The calibrated broadband spectra of individual targets are used to determine the number of organisms with resonance frequencies in the band. [Funding provided by the WHOI Ocean Twilight Zone Audacious Project and an NSF MRI grant.]

12:00

**1aAOb3. Broadband scattering from mesopelagic jellyfish.** Rachel E. Kahn (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 86 Water St. MS 12, Woods Hole, MA 02543, rkahn@whoi.edu) and Andone C. Lavery (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

By linking surface waters to the deep ocean, the mesopelagic zone (200–1000 m) plays crucial ecological roles in biogeochemical cycling, feeding apex predators, and sustaining a massive daily migration of organisms. The importance of this expansive ecosystem cannot be fully understood without accurately estimating its biomass. Several recent studies have estimated the biomass of mesopelagic fishes, yet the biomass of zooplankton is far less understood. The towed vehicle "Deep-See" was designed to obtain in situ acoustical (1–410 kHz), optical, and environmental measurements of mesopelagic fish and invertebrates. Dense patches of jellyfish were observed in images collected by Deep-See in the New England slope sea during Summer 2019, revealing the presence of gelatinous biomass potentially masked by more strongly scattering organisms (e.g., fish, krill) in narrowband shipboard acoustic surveys. To quantify this "hidden" biomass acoustically will require understanding the physics of sound scattered by these organisms. Using a full 3-D scan of an individual, and simpler approximate shapes, the Distorted Wave Born Approximation (DWBA) was used to model the target strength of the common mesopelagic jellyfish *Solmissus*. Images were used to predict the volume scattering spectrum of an observed *Solmissus* patch and compared with acoustical measurements.

12:20

**1aAOb4. Seasonal and annual variability in daily vertical migration observed from acoustic Doppler current profiler derived backscatter at the Mid-Atlantic Bight shelf break.** Jacob S. Partida (Physical Oceanogr., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS #12, Woods Hole, MA 02543, jpartida@whoi.edu), Weifeng G. Zhang, Andone C. Lavery (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), and J Michael Jech (NEFSC, Woods Hole, MA)

75 kHz ADCPs stationed near the 450-meter isobath within the Ocean Observatories Initiative Coastal Pioneer Array capture consistent patterns indicative of daily vertical migration (DVM) in depth profiles of both mean volume backscatter (MVB) and the vertical component of velocity ($w$). While the presence of DVM patterns is consistent throughout 2018 and 2019, attributes of MVB and $w$ profiles are subject to variability within and across different seasons and years. This variability may be influenced by abiotic factors characteristic of the Mid-Atlantic Bight region such as the disruption of typical spring time temperature-salinity structures via frontal instability or the episodic introduction of warm-core rings. Additionally, biological community composition in the region may be reflected in MVB and $w$ profiles given zooplankton survey data and knowledge of seasonal biophysical dynamics in the region, along with measured variability of other environmental properties observed by the Pioneer Array including dissolved oxygen concentration, chlorophyll-a concentration, and photosynthetically available radiation. Linking all of these factors, driven by long-term, stationary acoustical observations, can provide a useful framework for more directly assessing the impacts of aberrational physical phenomena on regional biological patterns.

## Session 1aBAa

## Biomedical Acoustics and Signal Processing in Acoustics: Death to Delay and Sum: Advanced Beamforming I

Kevin J. Haworth, Cochair
*Internal Medicine, University of Cincinnati, 231 Albert Sabin Way, CVC 3939, Cincinnati, OH 45267-0586*

Kenneth B. Bader, Cochair
*Department of Radiology, University of Chicago, 5835 South Cottage Grove Ave., MC 2026, Q301B, Chicago, IL 60637*

**Chair's Introduction—9:30**

### *Invited Papers*

**9:35**

**1aBAa1. Neural networks as an applied tool for ultrasound beamforming and image reconstruction.** Jeremy Dahl (Radiology, Stanford Univ., 3155 Porter Dr., MC 5483, Palo Alto, CA 94304, jjdahl@stanford.edu) and Dongwoon Hyun (Radiology, Stanford Univ., Palo Alto, CA)

In recent years, the computer science community has made available its artificial intelligence tools to the public at-large. Freely available online courses, software libraries, and databases have made it easy to incorporate artificial intelligence, and specifically machine learning, into many research areas. In medical imaging, this has led to a proliferation of machine learning-based research methodologies in image reconstruction tasks, whether appropriate or not. In this lecture, we review machine learning as a tool for ultrasound beamforming. We discuss the selection of an appropriate problem to be solved with machine learning, survey common and popular implementations of neural networks, and describe best practices and common pitfalls. We visit a few misconceptions about machine learning in ultrasound imaging and examine the recent literature in the field of ultrasound beamforming and image reconstruction.

**9:55**

**1aBAa2. Application-specific pulse-echo ultrasound image reconstruction using neural networks.** Dongwoon Hyun (Radiology, Stanford Univ., 3155 Porter Dr., Palo Alto, CA 94304, dongwoon.hyun@stanford.edu), Leandra Brickson, Lotfi Abou-Elkacem, Rakesh Bam, Carl Herickhoff, Kevin Looby, and Jeremy Dahl (Radiology, Stanford Univ., Palo Alto, CA)

Deep neural networks (DNNs) have recently emerged as powerful function approximators that can learn to transform given inputs into desired outputs when provided with enough training samples. Ultrasound B-mode images are traditionally formed using the delay-and-sum beamformer to display the echogenicity (i.e., backscattering strength) of the medium. Advanced beamformers are often designed to improve indirect metrics of image quality such as lesion detectability and point target resolution. Here, we describe how DNNs can be trained directly to minimize errors in the output images as compared to the ground truth targets and present recent work with DNNs for two specific applications. A simple convolutional DNN was trained to accurately estimate echogenicity for B-mode imaging using Field II simulations; the DNN produced speckle-reduced B-mode images of data acquired in simulations, phantoms, and *in vivo*. Similarly, a similar DNN was trained to detect targeted microbubbles for ultrasound molecular imaging; the DNN nondestructively detected microbubbles with similar performance to a state-of-the-art destructive imaging technique while enabling real-time molecular imaging.

**10:15**

**1aBAa3. Ultrasound beamforming using deep networks.** Brett Byram (Vanderbilt Univ., 2301 Vanderbilt Pl., Nashville, TN 37235, brett.c.byram@vanderbilt.edu), Jaime Tierney, and Matt Berger (Vanderbilt Univ., Nashville, TN)

Medical ultrasound exams contain substantial diagnostic information, but in many patients this information is obscured by various sources of degradation. Previously, we demonstrated that physical-model based beamforming methods could improve ultrasound image quality. This was important because it demonstrated that beamforming could be posed as a nonlinear regression problem, indicating that deep neural networks (DNNs) might also represent a beamforming solution. Based on this, we began developing such DNNs for ultrasound beamforming and showed that they did indeed lead to consistent improvements in simulation, phantom and *in vivo* data. For this effort, we considered several aspects of DNN training, including loss functions, optimization strategies, architectures and training data. The nature of the training data and loss functions exhibited the most substantial impact on performance. Acquiring appropriate training data is challenging because there are no existing curated data sets for beamforming, so we relied heavily on simulations, which can be carefully controlled to enable precise separation of signal and clutter. We also considered training with data acquired from phantoms where the signal and clutter could be easily separated. With respect to loss functions, we considered classic DNN methods, but we also recently developed dedicated, ultrasound motivated loss functions to good effect.

10:35

**1aBAa4. Deep learning the sound of light to guide surgeries.** Muyinatu Bell (Johns Hopkins Univ., 3400 N. Charles St., Barton 208, Baltimore, MD 21218, mledijubell@jhu.edu)

Photoacoustic imaging utilizes light and sound to make images by transmitting laser pulses that illuminate regions of interest, which subsequently absorb the light, causing thermal expansion and the generation of sound waves that are detected with conventional ultrasound transducers. The Photoacoustic and Ultrasonic Systems Engineering (PULSE) Lab is developing novel methods that use photoacoustic imaging to guide surgeries with the ultimate goal of eliminating surgical complications caused by injury to important structures like major blood vessels and nerves that are otherwise hidden from a surgeons immediate view. This invited presentation will summarize our recent work to learn from the physics of sound propagation in tissue and develop acoustic beamforming algorithms that improve image quality, using state-of-the-art deep learning methods. These deep learning methods hold promise for robotic tracking tasks, visualization and visual servoing of surgical tool tips, and assessment of relative distances between the surgical tool and nearby critical structures (e.g., major blood vessels and nerves that if injured will cause severe complications, paralysis, or patient death). Impacted surgeries and procedures include neurosurgery, spinal fusion surgery, hysterectomies, cardiac interventions, and biopsies.



MONDAY MORNING, 7 DECEMBER 2020 9:30 A.M. TO 11:00 A.M.

## Session 1aBAb

## Biomedical Acoustics: General Biomedical Acoustics: Elastography I

Gianmarco Pinton, Chair
*Biomedical Engineering, University of North Carolina at Chapel Hill and North Carolina State University, 116 Manning Drive, Mary Ellen Jones Room 9212A, Chapel Hill, NC 27599*

**Chair's Introduction—9:30**

### *Contributed Papers*

9:35

**1aBAb1. A time-aligned plane wave compounding method for high frame rate shear wave elastography.** Margherita Capriotti (Radiology, Mayo Clinic, 200 First St. SW, Rochester, MN 55905, capriotti.margherita@gmail.com), James Greenleaf (Physiol. and Biomedical Eng., Mayo Clinic, Rochester, MN), and Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN)

Shear wave elastography (SWE) is an established ultrasonic imaging technique, that provides quantitative measurements of tissue mechanical properties. This technique relies on the combination of acoustic radiation force push beams to generate propagating shear waves and ultrafast imaging, which allows high two-dimensional imaging frame rates to capture the induced motion. Utilizing coherent plane wave compounding (PWC) can improve signal-to-noise ratio (SNR) of acquired data. While the image quality improves, the pulse repetition frequency (PRF) is reduced, reaching limiting bandwidths to perform SWE. To address these challenges, a method for making SWE measurements with high PRF and SNR is presented. The idea follows the Time Aligned Sequential Tracking method, proposed to allow SWE with enhanced SNR in conventional clinical ultrasound scanners. The time alignment process is applied here to the full field-of-view, acquired by using plane waves transmitted at different angles. In this way, the full PRF and SNR gains are obtained. Results are shown from experiments in tissue-mimicking phantoms, where motion and extracted shear wave velocity are compared for the traditional and proposed data processing schemes. Preliminary results on arterial phantoms are also discussed, to emphasize the beneficial effects of the method in dispersive geometries.

9:55

**1aBAb2. Shear wave elastography for skeletal muscle diagnostics.** Timofey Krit (Dept. of Acoust., Faculty of Phys., Moscow State Univ., Leninskie Gory 1/2, Moscow 119991, Russian Federation, timofey@acs366.phys.msu.ru), Arina Ivanova (Dept. of Acoust., Faculty of Phys., Moscow State Univ., Moscow, Russian Federation), and Yuly Kamalov (Russian Sci. Ctr. of Surgery named after academician B.V. Petrovsky, Moscow, Russian Federation)

Shear modulus in the biceps of volunteers was measured with standard ultrasound equipment, the at loads from 0 to 50 N. The measurements were carried out by shear wave elastography in the clinic according to the medical protocol. The volunteer held a sport weight of known mass to load the bicep. Shear wave was excited in the muscle at a given depth by an ultrasonic sensor. The shear wave velocity was recorded in the specific point, which was determined by the position of the sensor. The shear modulus of the muscle fibers measured by elastography increases from 10 to 60 kPa with the increase of the load and returns to 10 kPa 1 min after load is removed. The maps of the shear modulus distribution were measured in the areas around the points of the shear velocity measurements. Research was funded by the grant from the Russian Science Foundation (Project No. 19-72-00086).

**10:15**

**1aBAb3. A noninvasive method to estimate tissue stress-strain relationships using quasi-static and shear wave ultrasound elastography.** Yuqi Wang (Dept. of Radiology, Mayo Clinic, 200 First St. SW, Rochester, MN 55902, wang.yuqi@mayo.edu) and Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN)

Current clinically used ultrasound elastography is obtained when tissue undergoes small deformation (<2% strain). However, small deformation may not be large enough to characterize tissues that exhibit similar elasticity properties to normal tissue at small deformation but behave differently from normal tissue at large deformation (>10% strain). Therefore, tissue elasticity properties measured at large deformation may provide better diagnostic differentiation. Combining the quasi-static and shear wave ultrasound elastography methods, we propose a noninvasive method to estimate the nonlinear elasticity property of tissue. To test this method, cubic tissue-mimicking phantoms were constructed. We progressively compressed the phantom to produce a series of stepwise stress states. At every stress state, we conducted shear wave measurements in which two acoustic radiation force pushes generated shear waves, which were measured using ultrafast imaging in order to estimate the local shear modulus. For every compression step, we tracked incremental tissue displacements between two successive frames and then accumulated these displacements for all steps to obtain the cumulative displacements and strains. Using the shear modulus and cumulative strain images we calculated the stress map under an incompressibility assumption. Finally, we obtained the local stress-strain curve that reflected the nonlinear elasticity of the tested material.

**10:35**

**1aBAb4. The effects of dispersion and noise on time-of-flight estimates of the shear wave speed in viscoelastic media.** Luke M. Wiseman (Elec. and Comput. Eng., Michigan State Univ., Michigan State University, East Lansing, MI, wisemanl@msu.edu), Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN), and Robert J. McGough (Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI)

Shear wave elasticity imaging applies an acoustic radiation force to generate shear waves, where the shear wave speed is often estimated with time-of-flight calculations. To characterize the errors in time-of-flight estimates of the shear wave speed, three-dimensional simulated shear wave particle velocities and shear wave particle displacements are computed with time-domain Green's functions for a Kelvin-Voigt model. Estimated shear wave speeds are obtained from cross correlations and time-to-peak calculations, and the results demonstrate the effects of time-domain dispersion and noise on estimates of the shear wave speed. Time-domain dispersion of the propagating shear wave produces large errors in estimates performed with shear wave particle velocities and/or cross-correlations close to the push beam, and in the absence of noise, the errors in the shear wave speed estimated at the focal depth diminish with distance from the push beam. However, shear wave attenuation also increases sensitivity to noise as the distance from the push beam increases. Time-to-peak and cross-correlation methods consistently achieve much smaller errors with shear wave particle displacements than with shear wave particle velocities. Furthermore, cross-correlations are much more robust with respect to noise, and larger values of the shear viscosity increase the sensitivity of each method to noise.

MONDAY MORNING, 7 DECEMBER 2020 — 11:15 A.M. TO 12:45 P.M.

# Session 1aBAc

## Biomedical Acoustics and Signal Processing in Acoustics: Death to Delay and Sum: Advanced Beamforming II

Kevin J. Haworth, Cochair
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Kenneth B. Bader, Cochair
*Department of Radiology, University of Chicago, 5835 South Cottage Grove Ave., MC 2026, Q301B, Chicago, IL 60637*

**Chair's Introduction—11:15**

### *Contributed Paper*

**11:20**

**1aBAc1. 3-D photoacoustic tomography reconstruction with deep learning for vasculature imaging.** Steven Guan (Bioengineering, George Mason Univ., 4400 University Dr., Fairfax, VA 22030, Sguan2@gmu.edu) and Parag V. Chitnis (Bioengineering, George Mason Univ., Fairfax, VA)

Photoacoustic tomography (PAT) is a hybrid imaging modality capable of acquiring high contrast and resolution images of optical absorption at depths greater than traditional optical imaging techniques. Practical considerations with instrumentation limit the number of acoustic sensors and their "view" of the imaging target, which result in image reconstruction artifacts degrading image quality. Iterative reconstruction methods can be used to improve image quality but are computationally expensive, especially for 3-D PAT over large imaging volumes. Deep learning has emerged as an efficient alternative and is capable of achieving state-of-the-art performance. In this work, we compare the 3-D fully dense UNet convolutional neural network with the widely used time reversal reconstruction method. Simulated acoustic data was generated using the *K*-wave MATLAB toolbox (225 detectors, planar geometry). A public database containing 50 patient's whole-lung CT scans was used to create training ($n = 40$) and testing ($n = 10$) data. The training and testing datasets are comprised of 500 and 50 lung vasculature volumes sampled from their respective whole-lung CT scans. Using the structural similarity index, the proposed deep learning method ($0.87 \pm 0.04$) is shown to be superior to time reversal ($0.38 \pm 0.11$).

## *Invited Papers*



**11:40**

**1aBAc2. Spectrally resolved passive acoustic imaging.** Costas Arvanitis (Dept. of Biomedical Eng., School of Mech. Eng., Georgia Inst. of Technol., 311 Ferst Dr. Northwest, Atlanta, GA 30332, costas.arvanitis@gatech.edu)

Ultrasound in combination with circulating microbubble contrast agents has emerged as a promising modality for therapy and imaging of brain diseases. Despite progress, the need for spatio-temporal mapping and control of the cerebrovascular microbubble dynamics necessitates the development of fast and spectrally resolved trans-skull imaging methods. In this talk, we will present our recent efforts towards establishing such methods. First, we will introduce a general methodology using the angular spectrum (AS) approach in a heterogeneous medium, which is based on a numerical marching scheme to approximate the full implicit solution, and reveal its functionality for trans-skull passive acoustic imaging (70% source localization error reduction from 2.89 ± 1.76 mm to 0.68 ± 52 mm). Then, we will demonstrate the ability of AS in combination with a nonlinear state controller that uses specific frequency bands of the microbubble acoustic emissions (harmonic, ultra-harmonic, etc.) to attain stable cavitation activity for tens of seconds. Finally, we will present our progress towards adapting the AS method for rapid visualization of vessel structures with diameters of a few hundreds of microns. The superior image quality and computation efficiency (>100-fold faster from time-domain delay-and-sum methods) of AS offers unique opportunities for spectrally resolved passive acoustic imaging and image guided therapy of human disease.

**12:00**

**1aBAc3. Weight for it… adaptive beamformers in passive acoustic mapping for cavitation imaging.** Michael Gray (Inst. of Biomedical Eng., Univ. of Oxford, Oxford OX3 7DQ, United Kingdom, michael.gray@eng.ox.ac.uk), Christian Coviello (OxSonics Therapeutics, Oxford, United Kingdom), Miklos Gyongy (Pazmany Peter Catholic Univ., Budapest, Hungary), Erasmia Lyka, Catherine Paverd (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), Calum Crake (OxSonics Therapeutics, Oxford, United Kingdom), Delphine Elbes (OrthoSon Ltd., Oxford, United Kingdom), Cameron Smith, and Constantin Coussios (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom)

As the applications of therapeutic ultrasound expand across neurological, oncological and musculoskeletal medicine, monitoring of cavitation activity offers the potential for non-invasive treatment guidance and optimization of safety and efficacy. When implemented with a distributed set of sensors, simple delay and sum beamformers can be effective for cavitation source detection, but localization and quantification capabilities may be hindered when using conventional diagnostic ultrasound arrays. More generally, when observations of the emitted field are limited by the quantity, bandwidth, and geometric distribution of the receivers employed, and/or are subject to uncertainties in array calibration and signal propagation path, ã robust' data-adaptive beamformers can offer considerable improvements in the estimation of emission source properties. This paper reviews the concepts of adaptive beamformers—wherein outputs are calculated using data-dependent receiver weights—and their application to cavitation monitoring. Examples are presented for two such beamformers used in our laboratory, demonstrating their enhanced imaging capabilities, their relative computational cost, and the physical interpretation of their optimized weights. Finally, we discuss clinical translation of these techniques, critical factors in quantitative data interpretation, and how beamformed data can be used to estimate and correct for *in vivo* propagation effects.

## *Contributed Paper*

**12:20**

**1aBAc4. Passive acoustic mapping utilizing compressed-domain processing for real-time monitoring of cavitation-enhanced drug delivery.** Calum Crake (OxSonics Therapeutics, The Magdalen Ctr., Robert Robinson Ave., Oxford OX4 4GA, United Kingdom, calum.crake@oxsonics.com), Paul Boulos, Maura Power, Edward Ellis, Florian Monnier, Alessandro Polcaro (OxSonics Therapeutics, Oxford, United Kingdom), Richard Kozick (Elec. & Comput. Eng. Dept., Bucknell Univ., Lewisburg, PA), and Christian Coviello (OxSonics Therapeutics, Oxford, United Kingdom)

Ultrasound-induced cavitation has been shown to improve the delivery of a range of therapeutics including small-molecule drugs, oncolytic viruses and immunotherapies, potentially enhancing their delivery and reducing their toxicity for treatment of solid tumours. Real-time monitoring methods such as passive acoustic mapping (PAM) may be employed to map cavitation activity during treatment. However, while advances in acquisition hardware facilitate outstanding monitoring capability, transfer, storage and processing of the resulting data remains a challenge. Previously we investigated the use of compressed sensing (CS) techniques to sparsely reconstruct full-rate time-series array data for subsequent conventional PAM. Here we extend this approach to directly perform PAM using the compressed-domain (or CS) data. In our approach the CS data is obtained from projections that jointly compress the full-rate array data. The PAM image is computed directly from the CS data using a sparse matching pursuit algorithm, with a dictionary formed from a discretized model using a reference sensor. Results showing accurate localization of cavitation activity using CS-PAM are obtained with simulated and *in-vivo* (porcine) measured data but processing only 1% of the full-rate data. Finally, common issues with voxel grid density that affect source localization in CS-PAM and conventional PAM are highlighted.

# Session 1aBAd

## Biomedical Acoustics: General Biomedical Acoustics: Elastography II

Gianmarco Pinton, Chair
*Biomedical Engineering, University of North Carolina at Chapel Hill and North Carolina State University, 116 Manning Drive, Mary Ellen Jones Room 9212A, Chapel Hill, NC 27599*

**Chair's Introduction—11:15**

### *Contributed Papers*

**11:20**

**1aBAd1. Guided wave inversion for arterial stiffness.** Tuhin Roy (Civil Eng., North Carolina State Univ., 3535 Ivy Commons Dr., Apt. 302, Raleigh, NC 27606, troy@ncsu.edu), Matthew W. Urban, James Greenleaf (Dept. of Radiology, Mayo Clinic, Rochester, MN), and Murthy Guddati (Civil Eng., North Carolina State Univ., Raleigh, NC)

Shear Wave Elastography (SWE) has been used to estimate arterial stiffness, an important biomarker for early cardiovascular diseases. In arterial SWE, the carotid artery is excited with acoustic radiation force and the observed wave propagation characteristics are used to estimate the arterial stiffness. We present an efficient guided wave inversion procedure, where the elastic modulus of the arterial wall is estimated by matching wave dispersion from experiments and simulation. Central to the proposed approach is an efficient forward model that computes the dispersion relation for the incompressible, immersed cylindrical tube; the model achieves high efficiency by combining semi-analytical finite element methods, selective reduced integration, and perfectly matched discrete layers. We utilize this forward model in an optimization framework to develop a streamlined methodology to estimate the modulus of the arterial wall. In this talk, we present the formulation of the method as well as validation using phantom experiments on 10 rubber tubes (VytaFlex-10), followed by preliminary work on inverting for the viscoelastic modulus.

**11:40**

**1aBAd2. Pilot studies on zebrafish echocardiography and zebrafish ultrasound vibro-elastography.** Xiaoming Zhang (Mayo Clinic, 200 1st St. SW, Rochester, MN 55905, zhang.xiaoming@mayo.edu), Alex Zhang, and Xiaolei Xu (Mayo Clinic, Rochester, MN)

Zebrafish are being increasingly used as animal models for human diseases such as cardiomyopathy and neuroblastoma because zebrafish have a nearly fully sequenced genome to humans. One pilot study is to develop zebrafish echocardiography. Because the heart of a zebrafish is small, a high frequency ultrasound system (50–80 MHz) is used for imaging. B-Mode imaging is analyzed for quantification of ejection fraction (EF), end-systolic volume (ESV), and end-diastolic volume (EDV). Pulse-wave Doppler (PWD) is used to calculate the E/A ratio (the peak velocities of E and A waves) and myocardial performance index (MPI). Techniques for improving image quality and repeatability of zebrafish echocardiography are discussed. Another pilot study is to develop zebrafish ultrasound vibro-elastography (ZUVE) for measuring the elastic properties of zebrafish. An adult female zebrafish is anesthetized for 3 min. A 0.1-s gentle harmonic vibration is generated on the tail of a zebrafish using a sphere tip indenter with a 3 mm diameter. Shear wave propagation in the zebrafish is measured using a high frequency 18 MHz ultrasound linear array probe. We conclude that ZUVE is safe, fast, and noninvasive and feasible for testing the elastic properties of zebrafish.

**12:00**

**1aBAd3. A fast and efficient ultrasound tomography using deep learning.** Sumukha Prasad (CSE, The Penn State Univ., 2115 Plaza Dr., State College, PA 16801, sub1206@psu.edu) and Mohamed Almekkawy (CSE, The Penn State Univ., State College, PA)

The potential of ultrasound tomography (UT) has been noticed to quantify the tissue acoustic properties for advanced clinical diagnosis. Conventionally, UT is a full-wave inversion (FWI) problem which is addressed through iterative methods formulated as an optimization problem. The existing iterative FWI methods are ill-posed and have poor non-linear mapping leading to low-resolution UT images with artifacts while also being computationally expensive. Recently, deep learning networks have proven their capabilities in solving many complex problems. We propose to incorporate the novelty of deep learning in UT, i.e., leveraging its potential to overcome ill-posedness and learn the direct representation mapping from the time-series sensor data to the spatial acoustical image of the region of interest. We built a deep learning model using an encoder-decoder architecture with a convolutional neural network (CNN). But CNNs are known to introduce artifacts so we employ a locally connected conditional random field (CRF) on top of the CNNs to enhance the UT image. The proposed CRF-CNN shows the feasibility of performing UT directly with deep learning with promising results with 9% more accuracy while the computational time is reduced to 3 min on synthetic ultrasound data.

## Session 1aCAa

# Computational Acoustics and Structural Acoustics and Vibration: Domain Truncation Techniques for Exterior Problems I

Anthony L. Bonomo, Cochair
*Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817*

Benjamin M. Goldsberry, Cochair
*Applied Research Laboratories at The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758*

**Chair's Introduction—9:30**

### *Invited Papers*

**9:35**

**1aCAa1. Review of truncation techniques for full and half space acoustic problems.** Mads Jakob Herring Jensen (COMSOL A/S, Diplomvej 373, Lyngby 2800, Denmark, mads@comsol.dk), Andres Garcia, Kirill Shaposhnikov, and Elin Svensson (COMSOL AB, Stockholm, Sweden)

Many acoustic problems involve modeling radiation or scattering phenomena, including transducer characterization and design, sonar applications, or studying the human HRTF. In all cases open non-reflecting conditions or truncation techniques need to be set up when using numerical simulation methods. Several methods exist to model the open domain depending on the used numerical technique. In this contribution we will present and study the performance of several of these methods under various conditions. This includes radiation boundary conditions, perfectly matched layers, hybrid finite element (FEM) and boundary element (BEM) setups, as well as other sponge or absorbing layer methods. These will be applied to both frequency domain and time domain problems. Of particular interest will also be the application of the truncation techniques to half space scattering problems encountered in the analysis of absorption or transmission characteristics. In these setups oblique angles of incidence exist, which leads to numerical challenges.

**9:55**

**1aCAa2. High order local absorbing boundary conditions for acoustic and elastic scattering.** Vianey Villamizar (Dept. of Mathematics, Brigham Young Univ., 275 TMCB, Provo, UT 84602, vianey@mathematics.byu.edu), Tahsin Khajah (Mech. Eng., Univ. of Texas at Tyler, Tyler, TX), Sebastian Acosta (Cardiology, College of Medicine, Houston, TX), Dane Grundvig (Mathematics, Brigham Young Univ., Provo, UT), Jacob Badger (Computational Eng. and Sci., Oden Inst., Austin, TX), and Otilio Rojas (Barcelona Supercomputing Ctr., Barcelona, Catalunia, Spain)

The formulation of high order local absorbing boundary conditions in terms of farfield expansions (FE-ABC) for time-harmonic waves is considered. Our previously constructed FE-ABC for single and multiple acoustic scattering are improved. Furthermore, we extend the FE-ABC to elastic scattering. By decomposing the vector elastic displacement in terms of scalar potentials, the Navier's equation of elasticity is reduced to a boundary value problem consisting of two Helmholtz equations coupled through their boundary conditions. As a result, the formulation of the elastic FE-ABC from the acoustic case becomes natural. In this work, we present some numerical results by coupling the FE-ABC with finite difference methods and a general curvilinear finite element method based on isogeometric analysis. We present our results for two and three dimensional acoustic and elastic scattering problems from obstacles of arbitrary shape.

**10:15**

**1aCAa3. A nonlocal boundary condition for domain truncation in frequency-domain Helmholtz problems.** Robert C. Kirby (Mathematics, Baylor Univ., One Bear Pl. #97328, Waco, TX 76798, robert_kirby@baylor.edu), Andreas Kloeckner (Comput. Sci., Univ. of Illinois at Urbana-Champaign, Urbana, IL), and Ben Sepanski (Mathematics, Baylor Univ., Waco, TX)

Numerical resolution of exterior Helmholtz problems requires some approach to domain truncation. As an alternative to approximate nonreflecting boundary conditions and invocation of the Dirichlet-to-Neumann map, we introduce a new, nonlocal boundary condition. This condition is exact and requires the evaluation of layer potentials involving the free space Green's function. However, it seems to work in general unstructured geometry, and Galerkin finite element discretization leads to convergence under some mesh constraints imposed by Garding-type inequalities. We will sketch this method, show how we have integrated Firedrake with pytential (a fast-multipole code), and discuss what issues this raises for extending Firedrake to work with more general nonlocal operators.

**10:35**

**1aCAa4. Complete radiation boundary conditions for acoustic waves.** Thomas Hagstrom (Mathematics, Southern Methodist Univ., PO Box 750156, Dallas, TX 75275-0156, thagstrom@smu.edu)

The radiation of energy to the far field is a central feature of acoustics. As such, efficient, convergent domain truncation algorithms are a necessary component of any software for simulating acoustic waves in the time domain. Complete radiation boundary conditions are, in our view, an ideal solution to this problem. In particular, they are provably spectrally convergent, depending on parameters which can be chosen automatically to guarantee any required accuracy; as they directly approximate both propagating and evanescent modes, the computational boundary can be placed arbitrarily close to scatterers or other inhomogeneities. In this talk we will outline the theory behind the method for scattering and waveguide problems with a homogeneous far field, discuss the simple implementation of the method via the solution of a coupled system wave equations in a thin double absorbing boundary layer, and consider its application in more complex settings including advective acoustics, stratified media, and random media. Numerical experiments in three space dimensions using high-order methods will be shown.

MONDAY MORNING, 7 DECEMBER 2020 — 11:15 A.M. TO 12:25 P.M.

## Session 1aCAb

## Computational Acoustics and Structural Acoustics and Vibration: Domain Truncation Techniques for Exterior Problems II

Anthony L. Bonomo, Cochair
*Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd, West Bethesda, MD 20817*

Benjamin M. Goldsberry, Cochair
*Applied Research Laboratories at The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758*

**Chair's Introduction—11:15**

### *Invited Paper*

**11:20**

**1aCAb1. Perfectly matched *discrete* layers for modeling unbounded domains.** Murthy Guddati (NC State Univ., 2501 Stinson Dr., NCSU-Civil Eng., Raleigh, NC 27695-7908, mnguddat@ncsu.edu)

The method of perfectly matched layers (PML) is based on the elegant idea of introducing attenuation while keeping impedance unaltered, thus resulting in effective absorption of waves without spurious reflections. Due to its simplicity and effectiveness, PML has been widely used over the past 25 years to model wave propagation in unbounded domains. However, an acknowledged challenge is that PML no longer preserves impedance once the domain and the PML region are discretized, leading to spurious reflection at the interface. This issue has been fixed with the help of so-called impedance preserving discretization, leading to the method of perfectly matched discrete layers (PMDL), developed more than a decade ago. The idea is based on the observation that midpoint integration *completely* eliminates the discretization error in impedance that arises from linear finite element discretization. Beyond eliminating reflections at the interface, impedance preserving discretization is shown to be crucial to achieving stability of absorbing boundary conditions in situations with differing signs of phase and group velocities, e.g., anisotropic media. This talk will be an exposition of the development of the PMDL by the author and his collaborators, staring with fundamental ideas impedance preserving discretization followed by more advanced ABCs for anisotropic and periodic media.

## *Contributed Papers*



11:40

**1aCAb2. Quantification of error bounds for model truncation.** Anthony L. Bonomo (Signatures Dept., Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd, West Bethesda, MD 20817, anthony.bonomo@gmail.com) and Matthew Craun (Signatures Dept., Naval Surface Warfare Ctr., Carderock Div., West Bethesda, MD)

Many computational techniques have been developed to approximate the Sommerfeld radiation condition for finite computational domains. Recently, two such techniques (the perfectly matched layer and an easier-to-implement alternative dubbed the "reasonably" matched layer) have been used to truncate extended structures, treating these structures as semi-infinite in an attempt to extend the applicability of the finite element method to higher frequencies. In order for this model truncation method to be truly useful, however, a means to predict the error due to this semi-infinite assumption must be readily available. In this presentation, the mean-value method of Skudrzyk is revisited and utilized to make *a priori* error estimates that bound the excursion of the true solution from that obtained using model truncation. Error bounds for both structure borne vibration and sound radiation are considered. [Work supported by ONR.]

12:00

**1aCAb3. A finite element-based mode-matching technique for computation of the scattered wave fields from spatiotemporally modulated elastic media.** Benjamin M. Goldsberry (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, bgoldsberry@utexas.edu), Samuel P. Wallen, and Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Previous work by the authors provided the derivation and implementation of a finite element approach to model nonreciprocal elastic wave propagation for media in which the material properties are functions of both space and time [Goldsberry *et al.*, *J. Acoust. Soc. Am.* **146**(1) (2019)]. That work employs a harmonic balance approach to calculate propagating modes in a periodic medium. The present work adapts the aforementioned finite element approach to compute the scattered elastic wave field from finite structures having spatiotemporally modulated material properties. Calculation of the scattered field requires a domain truncation technique that accounts for the radiating modes at each generated harmonic of the modulation frequency. We derive and implement a mode-matching boundary condition using a Galerkin framework, which weakly enforces the continuity of the stresses and displacements at the boundary between the computational and the exterior analytical domain. We show that this approach is similar to a Dirichlet-to-Neumann boundary condition. Finally, we demonstrate this domain truncation technique as a means to compute the reflected and transmitted elastic wave fields from a proposed nonreciprocal elastic wave circulator design. [Work supported by NSF EFRI.]

# Session 1aEA

## Engineering Acoustics: General Topics in Engineering Acoustics I

Thomas E. Blanford, Cochair
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Caleb F. Sieck, Cochair
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Michael R. Haberman, Cochair
*Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758*

**Chair's Introduction—9:30**

### *Contributed Papers*

**9:35**

**1aEA1. On an efficiency improvement of the sound fire extinguisher by the electro-magnet speaker.** Bong-young Kim (Commun. Eng., IT-College, Soongsil Univ., 369 Sangdoro, Dongjakgu, Seoul, South Korea, bykim8@ssu.ac.kr) and Myungjin Bae (Commun. Eng., Soongsil Univ., Seoul, South Korea)

The Sound Fire Extinguisher is a device that suppresses flames by outputting loud sounds. In order for the sound fire extinguisher to be used in the fire fighting field, it must be compact and lightweight. In this paper, the electro-magnets were used instead of the permanent-magnets of speakers to maximize the output while minimizing the increase in the weight and volume size. A general dynamic speaker uses a permanent-magnet to generate power to the voice coil. The method applied in this study increased the magnetic flux acting on the voice coil by applying electro-magnet instead of permanent-magnet. An experiment was conducted to compare the output efficiency of the speaker with the permanent-magnet applied and the speaker modified with the electro-magnet. As a result of the comparative measurement through the experiment, the existing permanent-magnet applied speaker outputs 150 W at 1.5 kg, while the speaker applied by changing to electro-magnet outputs 500 W at 2 kg. In order to increase the output of the speaker, the weight is increased by 33% but the output is increased by 233% by applying electro-magnet instead of permanent-magnet. This enabled the Sound Fire Extinguisher to achieve high power, miniaturization and light weight.

**9:55**

**1aEA2. On an improvement of low bass characteristics on the OLED flat panel speaker.** Sungtae Lee (Commun. Eng., Soongsil Univ., Wollong-myeon, 245 LG-ro, Gyeonggi-do, Paju-si 10845, South Korea, stlee1974@naver.com), Kwanho Park (Commun. Eng., Soongsil Univ., Gyeonggi-do, South Korea), and Myungjin Bae (Commun. Eng., Soongsil Univ., Seoul, South Korea)

Recently, researches using OLED flat panel displays as speakers have attracted attention. However, due to the local characteristics of the flat panel speaker, the bass characteristic below 500 Hz is insufficient. The flat panel loudspeaker drives several watts of power into the exciter attached to the back of the flat panel. The exciter of the flat plate has a characteristic that the sound is driven forward from the local region, so the characteristics of the bass region are deteriorated. Therefore, in this paper, the research was conducted to widen the sound by attaching the exciter to the rigid bar. It is a principle to compensate the bass region of the flat panel speaker using an array type exciter. In order to improve the bass sound, the rigidity bar is used instead of the array exciters to improve the manufacturing process and cost. Compared to the conventional flat panel speaker, when the rigid bar is attached, the bass bandwidth is improved to 200 Hz. In the future, it is necessary to minimize the effect of the rigid bar on the high-pitched area compared to using exciter array.

**10:15**

**1aEA3. Comparing traffic intensity estimates employing passive acoustic Radar and microwave Doppler Radar sensor.** Andrzej Czyzewski (Multimedia Systems, Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, andczyz@gmail.com)

The purpose of our applied research project is to develop an autonomous road sign with built-in radar devices of our design. In this paper, we show that it is possible to calibrate the acoustic vector sensor in such a way that it can be used to measure traffic volume and to count the vehicles involved in the traffic through the analysis of the noise emitted by them. Signals obtained from a Doppler radar are used as a reference source. Although the acoustical vector sensor (AVS), being the embodiment of acoustic radar, has a lower accuracy than Doppler radar in vehicle counting and it is not able to measure the vehicle speed with the same precision, it has some advantages over the Doppler sensor. Namely, it does not emit any signals, it is not susceptible to electromagnetic interferences and it allows for further analysis of audio signals, such as assessment of the road surface state (e.g., wet/dry). The acoustic radar we developed is a new proposition of the acoustic method for road traffic monitoring. In addition, our research allowed a comparison of the efficiency of both methods, i.e., microwave and acoustic ones. [Project No. POIR.04.01.04-0089/16 entitled: "INZNAK – Intelligent road signs with V2X interface for adaptive traffic controlling" is financed by the Polish National Centre for Research and Development from under the EU Operational Programme Innovative Economy.]

**10:35**

**1aEA4. Sound barriers from stoics to simplex and from optics to acoustics.** Giora Rosenhouse (Acoust., Swantech Ltd., 9 Kidron St., Haifa 3446310, Israel, giora@swantech.co.il)

The design of finite sound barriers near numerous noise sources and control points needs many calculations, especially when optimization is required. Since the problem involves geometry that combines generalized triangles and quadrilaterals, the use of simplex theory seems to be effective.

It was developed by Dantzig (1914–2005) for linear programming. This is one of the most influential algorithms in science, engineering and arts. In many situations, the basic simplex forms can support analysis of insertion loss (IL) of relatively complicated sound barrier forms by applying trigonometric relations of the basic simplices as suggested here. Specific known data of simplices (sides, angles, areas, volumes) solve additional unknowns of a whole 2-D or 3-D grid or part of it. Such calculations can apply vector analysis to define the angle size and the location of points in the simplical complex which is necessary as data for finding IL and other issues of environmental acoustics. We will show solutions of IL, including "Maekawa–Pierce triangle" in wide barriers, finite barrier analysis by combined triangles, combined passive—electro acoustic with passive anti-noise protection and Scholes-Sargent graphs of IL at the shadow zone of a bridge, expressed by the shadow angle of the control point.



MONDAY MORNING, 7 DECEMBER 2020 9:30 A.M. TO 11:00 A.M.

## Session 1aNSa

## Noise: General Topics in Noise I

S. Hales Swift, Chair
*Sandia National Laboratories, P.O. Box 5800, MS 1082, Albuquerque, NM 87123-1082*

**Chair's Introduction—9:30**

### *Contributed Papers*

**9:35**

**1aNSa1. Aerogel two ways: Towards ultralight noise absorbers.** Bhisham Sharma (Aerosp. Eng., Wichita State Univ., 1845 Fairmount St., Wichita, KS 67260, bhisham.sharma@wichita.edu), Dong Lin (Industrial and Manufacturing Systems Eng., Kansas State Univ., Manhattan, KS), Amrutha Dasyam (Aerosp. Eng., Wichita State Univ., Wichita, KS), and Guang Yang (Industrial and Manufacturing Systems Eng., Kansas State Univ., Manhattan, KS)

Aerogels are a class of synthetic materials that are derived by using supercritical drying to extract the liquid component of a gel and replacing it with gas. This process results in the formation of an ultra-lightweight nanoporous solid composed of up to 99.98% air by volume. In this talk, we explore the possibility of developing aerogel-based ultralightweight sound absorbers in two different forms. First, we focus on the acoustic characterization of aerogel granules. The absorption properties of four different aerogels with granule diameters ranging from 2 to 4000 $\mu$m and granule densities ranging 120 to 180 kg/m$^3$ are presented. Next, we discuss the additive manufacturing of aerogel-based sound absorbers using a freeze casting method. This novel method allows the fabrication of solid aerogel structures with controlled microstructural properties. The sound absorption and transmission loss properties of 3-D printed aerogel samples are measured using an impedance tube setup. We demonstrate that such structures can provide significantly higher absorption and transmission loss properties as compared with current passive noise reduction materials. [Work Sponsored by NASA EPSCoR.]

**9:55**

**1aNSa2. An acoustic evaluation of light weight granular porous materials for urban air mobility applications.** Bharath Kenchappa (Mech. Eng., North Carolina A&T State Univ., 1601 E Market St., Greensboro, NC 27411, bkenchappa@aggies.ncat.edu) and Kunigal Shivakumar (Mech. Eng., North Carolina A&T State Univ., Greensboro, NC)

Soon Urban Air Mobility (UAM) vehicles will become a reality. The UAM vehicles need to be safe, environmentally clean, quieter and affordable. Air vehicle noise and reduction of it is a major challenge in urban areas, where these vehicles are needed the most. This paper attempts to develop a lightweight noise absorbing material system using porous particles constructed such that one can have varying pore size, its distribution, surface roughness, tortuosity, and layering. Analytical modeling is used to develop material model consisted of special hollow microbubbles, aerogel granules and other lighter weight materials. These partials are bonded by coating with a very thin layer of binder and processed to produce different structural constructions. The target is to achieve at least 50% sound absorption and 15 dB transmission loss. Several designed material systems will be validated by impedance tube testing. Details of modeling, development of materials, test and test results will be discussed in the full paper and in the presentation.

**10:15**

**1aNSa3. Enhanced low frequency broadband dissipation using micro-capillary plates.** Teresa Bravo (Instituto de Tecnologias Fisicas y de la Informacion, Consejo Superior de Investigaciones Cientificas, Serrano 144, Madrid 28006, Spain, teresa.bravo@csic.es) and Cedric Maury (Aix Marseille Univ, CNRS, Centrale Marseille, Marseille, France)

Helmholtz-type resonance absorbers constitute noise control devices widely used in many areas that have attracted attention in the last years due to the advancement of acoustic metamaterials. Micro-perforated panels placed over a backing cavity work on the same principle at low frequencies. They can provide important absorption values but confined in a narrow frequency band. To overcome this limitation, unbacked configurations have been considered, but care has to be taken for a proper selection of their constitutive parameters. In this work, freestanding micro-perforated plates with holes diameters down to 10 micrometers and high perforation ratio are shown to be good candidates as wideband low-frequency sound absorbers. Several micro-capillary plates, classified as a function of the Knudsen number, are studied analytically and experimentally. Most of the porous microsystems that use gases work in slip-flow regime whose properties differ considerably from the classical continuum regime. Results showed that

unbacked micro-capillary plates can achieve absorption values greater than 0.7 up to 7 kHz with an absorption plateau above 0.85 up to 4 kHz under normal incidence. Dependence of their performance to the backing load is analyzed. They could be used as low-frequency noise dissipation devices with applications as calibrated anechoic terminations.

**10:35**

**1aNSa4. Experimental characterization of aero-acoustic performance of micro-perforated materials in a wind tunnel.** Cedric Maury (Aix Marseille Univ, CNRS, Centrale Marseille, Marseille, France), Teresa Bravo (Instituto de Tecnologias Fisicas y de la Informacion, Consejo Superior de Investigaciones Cientificas, Serrano 144, Madrid 28006, Spain, teresa.bravo@csic.es), Muriel Amielh, Daniel Mazzoni, and Laurence Pietri (Institut de Recherche sur les Phénomènes Hors Equilibres, Marseille, France)

Mitigating the propagation of low frequency noise in ducted flows represents a challenging task since wall treatments have often limited dimensions. Micro-perforated panels have been successfully used backed by a cavity constituting a bulk-reacting resonator or in combination with honeycomb for a locally reacting system. In this work, a cost-efficient methodology for the study and characterization of the aero-acoustic properties of flush-mounted micro-perforated resonators has been developed. Although Laser Doppler Velocimetry is a non-intrusive technique, it requires delicate instrumentation to obtain an estimation of the acoustic velocity and pressure fluctuations at several points over a wall liner. Here, the attenuation has been estimated from sound pressure level measurements performed with two nosecone microphones positioned along a vertical line centered on the resonator axis. Experiments have been performed in a wind tunnel in presence of a low-speed turbulent boundary layer of air fully developed over different samples made up of micro-perforated sheets or porous materials with surface roughness. The micro-perforates have been flush-mounted over a cylindrical cavity of depth 30 mm situated on the floor of the test section. The ability of bulk-reacting resonators for reducing the acoustic or flow-induced noise has been assessed in comparison with locally reacting treatments.

MONDAY MORNING, 7 DECEMBER 2020 — 11:15 A.M. TO 12:45 P.M.

## Session 1aNSb

## Noise: General Topics in Noise II

S. Hales Swift, Chair
*Sandia National Laboratories, P.O. Box 5800, MS 1082, Albuquerque, NM 87123-1082*

**Chair's Introduction—11:15**

### *Contributed Papers*

**11:20**

**1aNSb1. Comparison of wind turbine-only noise levels measured at operating wind farms in the United States to noise levels predicted using ISO 9613-2:1996.** Michael Hankard, Jeff Cerjan (Acoust., Hankard Environ., Verona, WI), Justin Bowers (Acoust., Hankard Environ., 211 E Verona Ave., Verona, WI 53593, jbowers@hankardinc.com), and Scott McElroy (Acoust., Hankard Environ., Verona, WI)

This paper describes robust and effective methods for the measurement of noise emissions from utility-scale wind farms and the analysis of the resulting data for determining compliance with noise level limits and standards. The methods call for primarily unattended monitoring of noise levels, wind speeds, and turbine operating conditions over many weeks, in some cases months, in order to capture the recurring maximum wind turbine-only noise level at a receptor (i.e., residence). This condition occurs when the nearest turbines are producing full acoustic output while noise from background sources is very low (particularly that from wind). Presented are the results of noise compliance measurement studies applying these methods conducted by Hankard Environmental, encompassing eight projects and 35 measurement locations in the Northeast, Midwest, and Plains regions of the United States (U.S.) The results of these measurements are used to assess the accuracy of the ISO 9613-2:1996 method in predicting wind turbine noise levels at residences. These results can be used by wind energy developers to accurately design projects to meet applicable limits and standards and give all parties involved in the permitting process a greater degree of confidence in predictions and greater clarity during compliance measurements.

**11:40**

**1aNSb2. Automatic classification and reduction of wind noise contamination in spectral data.** Mylan R. Cook (Phys. and Astronomy, Brigham Young Univ., N201 ESC, Provo, UT 84602, mylan.cook@gmail.com), Zachary T. Jones (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Kent L. Gee, Mark K. Transtrum (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Matt Calton (Blue Ridge Res. and Consulting, Asheville, NC), Shane V. Lympany, and Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC)

As part of a larger project to develop a machine-learning-based model for ambient soundscapes, an investigation was conducted to determine what portion of the spectral database used for model training is likely contaminated by wind noise, and to remove this contamination. Although outdoor wind screens were used in data collection, many of the hourly one-third octave spectra have a low-frequency level increase that is characteristic of wind noise contamination. This contamination, which can extend to frequencies above 100 Hz, limits the machine learning algorithm's ability to train on acoustically relevant low-frequency data. In an attempt to automate classification, various spectral parameters were investigated for their

relevance in evidencing wind contamination. Although other spectral parameters can have a small impact on results, the low-frequency spectral slope is used as the key indicator for whether or not data are contaminated by wind noise. Using this approach, an automated classifier was created, which leads to a wind noise contamination reduction algorithm that can significantly improve sound level calculation. [Funded by a U.S. Army SBIR.]

**12:00**

**1aNSb3. On analysis of factors for retaliation psychology of the Klaxon sound.** Zhixing Tian (Commun. Eng., IT-college, Soongsil Univ., 369 Sangdoro, Dongjakgu, Seoul, South Korea, tzx596227421@gmail.com) and myungjin bae (Commun. Eng., Soongsil Univ., Seoul, South Korea)

The number of cars is increasing, the road environment is more complicated, the traffic is congested, and klaxon sound is used more. At this time, driving retaliation caused by klaxon sound has become a serious social problem. Therefore, it is necessary to develop a new rhythm Klaxon sound that does not cause psychological stress and retaliation. In this paper, we developed the optimal rhythm Klaxon sound by finding the cause of stress, retaliation, and noise in acoustic and psychological aspects through comparison of existing Klaxon sound and rhythm sound. Retaliation in the brain waves of listeners was shown to be delta wave and beta wave, respectively, and the degree of surprise was obtained according to the listening distance and sound level. The sound was chosen to have a smaller value for retaliation compared to the simple sustaining and rhythmic repetition of the Clarkson sound. When the object test was performed, the rhythm Klaxon sound was reduced by $-15$ dB and the EEG was reduced by $-10\%$. In conclusion, the simpler the klaxon sound, the more stress and surprises people felt. Using rhythm patten sound reduced noise and increased interest in sound.

**12:20**

**1aNSb4. 3-D printed bulk absorbers.** Bhisham Sharma (Aerosp. Eng., Wichita State Univ., 1845 Fairmount St., Wichita, KS 67260, bhisham.sharma@wichita.edu), Kyle Wetter, and Will Johnston (Aerosp. Eng., Wichita State Univ., Wichita, KS)

We present results from our recent efforts on the fabrication and acoustic characterization of 3-D printed porous absorbers. The acoustic properties of porous structures are dependent on their cellular microstructural architecture. Thus, controlling their local and global cellular architecture can allow a significant control over the acoustic properties of porous materials. Our focus here is on two different kinds of porous absorbers—open-celled absorbers with novel surface topologies and functional gradients, and bio-inspired fibrous absorbers. First, we apply stereolithographic printing to fabricate open-celled absorbers with novel surface topologies. The effect of parameters such as porosity, relative density, surface topology, and through-thickness porosity gradients on the sound absorption behavior is elucidated. Second, we explore the possibility of fabricating fibrous absorbers using fused deposition modeling. While 3-D printing allows fabrication of complex geometries, printing thin fibrous parts is significantly more challenging. Here, we compare two different techniques of printing such structures and present the effect of the fibrous architecture on the acoustic properties of the structure. The obtained results show that 3-D printing offers a promising new avenue of designing porous structures with tailored acoustic properties.

MONDAY MORNING, 7 DECEMBER 2020 — 9:30 A.M. TO 11:00 A.M.

# Session 1aPAa

## Physical Acoustics: General Topics: Acoustic Characterization of Materials I

Blake E. Simon, Chair
*Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758*

**Chair's Introduction—9:30**

### *Contributed Papers*

**9:35**

**1aPAa1. Acoustic measurement of the flow resistivity of a rigid porous material via low frequency transmitted waves—Frequency approach.** Mustapha Sadouki (Département des Sci. de la Matière, Université de Khemis-Miliana, Rout Thenia el Had, Khemis-Miliana 44225, Algeria, mustapha.sadouki@univ-dbkm.dz), Nassima Ait Kaid, and Hanane Hassine (Département des Sci. de la matière, Université de khemis-Miliana, Khemis-Miliana, Algeria)

In this work, an acoustic characterization process is proposed to measure the flow resistivity of porous materials with rigid frame via transmitted waves at very low frequency. The equivalent fluid theory is considered. Two expressions of the transmission coefficient have been established in the Darcy regime. The first depends on the frequency, porosity and air flow resistivity, while the second is frequency independent and depends only on the air flow resistivity. The inverse problem is solved in the frequency domain. The inverted values of the air flow resistivity are determined using the two low-frequency expressions of the transmission coefficients. The tests are carried out using two industrial plastic foam samples frequently used in thermal and sound insulation. The reliability of the results obtained is discussed and compared with those given by conventional methods.

9:55

**1aPAa2. Sensitivity study of the acoustic parameters on the wave transmitted through a bilayer porous medium at very low frequencies.** Mustapha Sadouki (Département des Sci. de la Matière, Université de Khemis-Miliana, Rout Thenia el Had, Khemis-Miliana 44225, Algeria, mustapha.sadouki@univ-dbkm.dz), Nassiba Rahmoun, and Kenza Boudani (Département des Sci. de la Matière, Université de Khemis-Miliana, Khemis-Miliana, Algeria)

The objective of this work is to present a sensitivity study of very low frequency parameters on waves transmitted through a bilayer medium made of two different layers of a porous medium saturated with air. The equivalent fluid theory is considered. Viscous-inertial and thermal effects are taken into account by the dynamic tortuosity and dynamic compressibility given by the Johnson-Champoux-Allard (JCA) model. At very low frequencies, the dynamic tortuosity and compressibility depend only on the porosity and air flow resistivity. An expression of the transmission coefficient of a bilayer medium has been established in the frequency regime. This expression depends on the porosity and resistivity of each layer. The influence of the ð20% variation in these parameters on the transmitted signal was presented and discussed. This study was done in the frequency domain as well as in the time domain.

10:15

**1aPAa3. Determination of photoelastic properties of silicon crystals in visible range.** Farkhad Akhmedzhanov (Lab. of Thermophysics of Multiphase Systems, Inst. of Ion-plasma and Laser Technologies of the Acad. of Sci. of Uzbekistan, 33 Durmon yuli St., Tashkent 100125, Uzbekistan, akhmedzhanov.f@gmail.com)

As is well known, the acousto-optic interaction in absolutely transparent crystals is lacking. In this connection the greatest interest are presented the acousto-optic investigations in the optical range, in which the crystals are not transparent. In order to carry out the similar experiments, it is necessary to use a buffer crystal with a small coefficient of light absorption at the applied wavelength. In the present work the effective photoelastic constant of a silicon crystal was determined by the Bragg light diffraction method at a wavelength 632.8 nm. A gallium phosphide crystal sample was used as a buffer crystal because it is transparent in the visible range of the spectrum and has good acoustooptical quality. The research results have shown the possibility of determination of photoelastic properties of silicon crystals by the method of Bragg light diffraction in the visible range of the light. In such experiments, the intensity of diffracted light can be obtained by an order of magnitude greater than in crystals transparent in the visible region of the light, such as lithium niobate, paratellurite, or lead molybdate.

10:35

**1aPAa4. Adaptive model-based control of traveling wave in a dispersive impedance tube.** Yoav Vered (Mech. Eng., Technion - Israel Inst. of Technol., Dynam. Lab. Mech. Eng. Faculty, Technion, Haifa 3200003, Israel, syoavv@gmail.com) and Izhak Bucher (Mech. Eng., Technion - Israel Inst. of Technol., Haifa, Israel)

Impedance tubes are the preferred instrument for measuring the acoustic properties of specimens in fluidic mediums. It is shown that the accuracy of the estimation improves by manipulating the boundary impedance of the tube. By means of digital control, a wide range of boundary impedances can be realized showing greater flexibility than passive termination. This is crucial especially when regarding a wide frequency range. In the realistic case of an elastic waveguide, multiple dispersive modes propagate simultaneously at all frequencies. Boundary control should, therefore, account for the modes and their dispersive nature. In this paper, the tube boundary impedance is being tuned as desired by using two loudspeakers, one at each end. The suggested method accommodates the dispersive behavior by estimating, in real-time, a parametric reduced-order model using an adaptive algorithm. A nonlinear control law is employed, capable of tracking the desirable modal traveling wave ratio. An experimental case-study utilizing an air-filled impedance tube is described. The results demonstrate the proposed approach's ability to control the dynamics of the principal acoustic mode, enable to create any combination of standing or traveling waves along the tube.

## Session 1aPAb

## Physical Acoustics: General Topics: Elastic Wave Propagation I

John M. Cormack, Chair
*Department of Medicine, University of Pittsburgh, Pittsburgh, PA 15261*

**Chair's Introduction—9:30**

### *Contributed Papers*

**9:35**

**1aPAb1. Propagation and distortion of surface acoustic waves along the boundary of a bimodule medium.** Vladimir A. Gusev (Physical Faculty, Dept. of Acoust., Lomonosov Moscow State Univ., Leninsky Gory, Moscow 119991, Russian Federation, vgusev@bk.ru)

The propagation of surface acoustic waves at the boundary of an elastic bimodulus medium is described. The developed model of medium allows to describe acoustic phenomena with non-classical nonlinearity with amplitude-dependent threshold effects, caused by the presence of internal structure such as cracks or polymeric chains. The temporal profile of the surface wave is built. It is shown that this profile in the bimodulus medium is fundamentally asymmetric. This leads to the formation of the unipolar limit profile at large distances.

**9:55**

**1aPAb2. Damage size quantification in one-dimensional bar using nonlinear ultrasonics.** Pravinkumar R. Ghodake (Mech. Eng., Indian Inst. of Technol., Bombay, B-423, Hostel 14, Mumbai, Maharashtra 400076, India, mech7pkumar@gmail.com) and Salil S. Kulkarni (Mech. Eng., Indian Inst. of Technol., Bombay, Mumbai, Maharashtra, India)

The problem of finding the size of localized damage in a one-dimensional bar using nonlinear ultrasonics is explored in this study. The damaged region is considered as a material with both quadratic as well as cubic nonlinearity, considered separately, located between two linear elastic regions. Two different inverse problems are formulated depending on the type of material nonlinearity assumed for the damaged region. The inverse problem is formulated considering both the backscattered and transmitted waves which are functions of input frequency, linear wave velocity and damage size. The inverse problem is reformulated as an optimization problem in order to circumvent the problem of non-uniqueness of the solution. Gradient and non-gradient based methods are used to solve the optimization problem. The effectiveness of the proposed method is demonstrated numerically by solving the inverse problems by using synthetic data generated by first solving the forward problems. The robustness of the solution is also verified by adding noise to the synthetic data. It is observed that for the case of quadratically nonlinear damage, at least two independent experiments are needed to find the solution, but for cubically nonlinear case only one experiment is sufficient for most of the cases.

**10:15**

**1aPAb3. Nonclassical nonlinearity from dislocation dynamics in resonant vibrations of structures.** Aakash Khandelwal (Dept. of Mech. Eng., Mech. Eng., Michigan State Univ., 428 S. Shaw Ln., Rm. 2555, Eng. Bldg., East Lansing, MI 48824, khande10@egr.msu.edu) and Sunil Kishore Chakrapani (Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI)

The dislocation string model developed by Koehler, Granato and Lücke has been used to study the contribution of dislocations to the generation of higher harmonics and damping in propagating acoustic waves. The bowing, and subsequent breakaway of dislocation loops at sufficiently high stress amplitudes leads to a stress amplitude dependence of the acoustic nonlinearity and damping. The present work focuses on the development of a model which incorporates the effects of this amplitude dependence in resonant vibration of solid structures. The equation of motion for a harmonically forced nonlinear beam was derived following the classical plate theory, and assuming a material model showing quadratic nonlinearity with linear viscoelasticity. The dislocation contribution was used to express the coefficients in the equation of motion as forcing dependent parameters. The frequency response and resonant frequency shift of the nonlinear beam resulting from the developed model show a deviation from classical nonlinear behavior, including softening-hardening nonlinearity. The models developed here show that the dislocation contribution can be nonclassical in nature.

**10:35**

**1aPAb4. Stress formalism for elastic waves in anisotropic solids.** Christopher M. Kube (Eng. Sci. and Mech., The Penn State Univ., 212 Earth and Eng. Sci. Bldg, University Park, PA 16802, kube@psu.edu) and Andrew N. Norris (Mech. & Aerosp. Eng., Rutgers Univ., Piscataway, NJ)

Traditionally, the propagation of bulk elastic waves in anisotropic crystals or solids is governed by the displacement form of the elastodynamic wave equation. Phase velocities follow after transforming the wave equation into an eigenvalue problem, which gives the Christoffel equations, and solving for the roots of the characteristic polynomial. Alternatively, only a few researchers have considered a pure stress wave equation of motion to describe the propagation of waves in elastic solids. However, to our knowledge, the dual characteristic polynomial and its solutions based on a pure stress formalism has not been established previously. This presentation will outline the needed steps to reach and solve the dual characteristic polynomial. Phase velocity solutions follow from the formation of new principal invariants, which are functions of the elastic compliance constants and propagation directions. The dual formalism is proven consistent to the traditional displacement formalism through the use of sixth-rank Levi-Civitia identities. Lastly, possible extensions of this work will be highlighted.

# Session 1aPAc

## Physical Acoustics: General Topics: Elastic Wave Propagation II

John M. Cormack, Chair
*MDepartment of Medicine, University of Pittsburgh, Pittsburgh, PA 15261*

**Chair's Introduction—11:15**

### *Contributed Papers*

**11:20**

**1aPAc1. A new theory of slow dynamics: Mechanistic diffusion model.** James Bittner (Michigan Technolog. Univ., P.O. Box 22, Houghton, MI 49931, jbittner@mtu.edu) and John Popovics (Univ. of Illinois at Urbana/Champaign, Urbana, IL)

Wave propagation through damaged heterogeneous materials (e.g., stone, cement, metals) elicits a broad range of physical behaviors, which includes a transient nonlinear modulus softening known as *slow dynamics*. Slow dynamic behaviors may provide new capability for identifying the presence of damage in materials that are traditionally difficult to inspect. Yet, to date, no verified physical justification for the presence of slow dynamic behavior exists and no single model accounts for all experimental observations. In this work, we propose a new theory for slow dynamic behaviors based on a mechanistic multi-physics diffusion model, and we evaluate the theory using a new experiment. Our results demonstrate that the new theory provides a physical mechanism for slow dynamic behaviors that aligns well with existing experimental observations, and the experiment enables measurement of a new physical quantity that correlates with transient nonlinear modulus softening. These results can be used to create new techniques of measuring slow dynamics for improved detection of damage in heterogeneous materials.

**11:40**

**1aPAc2. Elastic softening of sandstone due to a wideband acoustic pulse observed by dynamic acousto-elastic testing.** John M. Cormack (Ctr. for Ultrasound Molecular Imaging and Therapeutics, Dept. of Medicine, Univ. of Pittsburgh, Pittsburgh, PA 15261, jmc345@pitt.edu), Thomas G. Muir, Charles M. Slack, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Laboratory measurements of finite-amplitude, wideband elastic pulse propagation in a thin bar of Texas moss sandstone have been reported [Muir *et al.*, *JASA* **147** (2020)] that utilized pulses generated by impact excitation with a pendulum hammer at one end of the bar. Each pulse is essentially unipolar in strain, with bandwidth of approximately 3 kHz and amplitude between 10 and 130 microstrain. Reduction of the sandstone elastic modulus by up to 17% resulted from propagation of the impact-generated pulse, as calculated from time-of-flight measurements of the pulse as it reflected between the ends of the bar. In the present work, elastic softening is observed locally using a Dynamic Acousto-Elastic Testing (DAET) configuration. During propagation of an impact-generated pulse, changes in the sandstone elastic properties are observed by tracking variations in the arrival time and amplitude of a continuous-wave 1 MHz P-wave signal that propagates perpendicular to the bar axis. Repetitive reflection of the pulse from the stress-free ends of the bar enables separate observation of effects from compressive and tensile strains. Reduction of the elastic modulus of up to 20% is observed, with almost all of the softening occurring during the tensile phase of the impact-generated pulse. [a]Deceased.

**12:00**

**1aPAc3. Nonlinear torsional wave propagation in a thin circular rod.** John M. Cormack (Ctr. for Ultrasound Molecular Imaging and Therapeutics, Dept. of Medicine, Univ. of Pittsburgh, Pittsburgh, PA 15261, jmc345@pitt.edu) and Mark F. Hamilton (Walker Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Coupled nonlinear wave equations are obtained for the special case of wave motion in a thin rod with circular cross-section resulting from a large-amplitude torsional source. Lagrangian mechanics are employed to obtain the wave equations from expressions for the kinetic and potential energy densities in the rod due to an assumed displacement field. Longitudinal waves are generated at quadratic order in strain by the torsional wave through inertial (centrifugal) and elastic forces. Nonlinear effects on the torsional wave occur at cubic order in strain, due to both interaction with the longitudinal wave and strain-hardening shear nonlinearity. While the torsional wave propagates without dispersion, the longitudinal wave is subject to leading-order effects of geometrical dispersion when its wavelength is comparable to the rod radius. The case of weak nonlinearity is analyzed, for which the torsional wave propagates without nonlinear distortion while finite-amplitude effects generate a longitudinal wave. An analytical solution reveals that inertial and elastic forces resulting from a harmonic travelling torsional wave work against each other in the generation of the longitudinal second harmonic. Numerical solutions for transient source motion highlight interaction between the torsional and longitudinal modes near the source, and effects of geometrical dispersion on the longitudinal wave.

**12:20**

**1aPAc4. Experimental comparison of acoustic and seismic excitation methods for buried containers.** James Sabatier (Univ. of MS, 16 County Rd. 3062, Oxford, MS 38655, sabatier@olemiss.edu)

Acoustic and seismic sources have been used to excite vibrations in buried containers. However, there have not been experimental comparisons of these source types to determine the target vibration excitation efficiency. Comparison of the acoustically and seismically induced vibration signals on the ground surface directly over and away from the buried container is a straightforward approach to quantify the efficiency. A finite element modeling [Muir *et al.*, "Comparison of acoustic and seismic excitation, propagation, and scattering at an air-ground interface containing a mine-like inclusion," *J. Acousti. Soc.. Am.* **135**(1), 49–57 (2014)] of a buried target has quantified this efficiency by comparing calculated vibrational levels for acoustic and seismic sources before and after the "container" is put in place in the model. The modeling concludes the acoustic source is 10 dB greater than the seismic source. An experimental effort to compare the excitation efficiency of acoustic and seismic excitation techniques is reported in this paper. The sources are seismic and acoustic and include continuous wave and impulsive excitation at short and long ranges.

## Session 1aPAd

## Physical Acoustics: General Topics: Ultrasonics

Samuel P. Wallen, Cochair
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Kevin M. Lee, Cochair
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**Chair's Introduction—11:15**

### *Contributed Papers*

**11:20**

**1aPAd1. Numerical study of reflectivity signals with diffracting airborne ultrasound beams.** Alejandro Ortega-Aguilar (Instituto de Ciencias Aplicadas y Tecnología, Universidad Nacional Autónoma de México, Apartado Postal 70-186, Ciudad de México 04510, Mexico, alorag3.14159@gmail.com), Roberto Velasco-Segura (Instituto de Ciencias Aplicadas y Tecnología, Universidad Nacional Autónoma de México, Ciudad Universitaria, Mexico City, Mexico), and Augusto García-Valenzuela (Instituto de Ciencias Aplicadas y Tecnología, Universidad Nacional Autónoma de México, Ciudad de México, Mexico)

In this work, we investigate an efficient method for calculating the reflected and transmitted acoustic fields from diffracting ultrasound beams incident to a gas/gas interface. The study is motivated by the application to gas sensors of an experimental setup consisting of air and CO2 separated by a thin membrane and a 40 kHz acoustic incident beam at oblique angles. The numerical method consists of performing a Plane Wave Expansion (PWE) of the incident beam using standard Fourier transform methods. Expressions for the reflected and transmitted acoustic fields are obtained in terms of two-dimensional integrals of simple Kernels. These can be performed numerically in a straightforward way. The main approximation in this method comes from guessing the acoustic pressure field at a plane just in front of the ultrasonic emitter. We compare calculations with the PWE method with Finite Element Method simulations in a 2-D geometry, finding good agreement. Among the advantages of the PWE method we have that there is no need to calculate the whole acoustic fields in a large volume of many cubic wavelengths, while still considering rigorously diffraction effects. We will present predictions for practical airborne ultrasound reflection measurements of interest in developing novel acoustic gas sensors.

**11:40**

**1aPAd2. Ultrasonic differential travel time flow measurement in liquids with suspended granular solids.** Max Deffenbaugh (Aramco Res. Center–Houston, Aramco Services Co., 17155 Park Row, Houston, TX 77084, max.deffenbaugh@aramcoamericas.com), Timothy Thiel (Aramco Res. Center–Houston, Aramco Services Co., Houston, TX), Chinthaka Gooneratne (EXPEC Adv. Res. Ctr., Saudi Aramco, Dhahran, Saudi Arabia), and Albert J. Williams (Woods Hole Oceanographic Inst., Woods Hole, MA)

Flow rates in homogeneous liquids can be determined by simultaneously transmitting ultrasonic pulses in opposite directions along the same path and measuring the travel time difference between the pulses travelling with versus against the flow. We discuss the feasibility of extending this measurement technique to flows of a liquid with suspended granular solids. Two new factors limit the accuracy of the flow measurement for this mixture. First, the solids cause spreading and attenuation of the ultrasonic pulse, reducing the travel time measurement accuracy. Second, the flow moves slightly over the travel time of the pulse, such that the oppositely directed pulses do not encounter exactly the same medium. We present lab experiments on a liquid with suspended granular solids similar to the drilling fluids which are circulated through the bit during the drilling of oil and gas wells. The flow measurement accuracy is predicted as a function of the wavelength, grain size, volume fraction of solids, path length and flow velocity.

**12:00**

**1aPAd3. Multi-particle assembling patterning by acoustic forces and scattering dissipation.** Tianquan Tang (Dept. of Mech. Eng., The Univ. of Hong Kong, Pokfulam Rd., Hong Kong, Lab. for Aerodynamics and Acoust., Zhejiang Inst. of Res. and Innovation, The University of Hong Kong, Hong Kong, Hong Kong 999077, Hong Kong, tianquan@connect.hku.hk) and Lixi Huang (Mech. Eng., The Univ. of Hong Kong, Hong Kong, Hong Kong)

There has been growing interest in exerting the radiation forces to trap and cluster the randomly distributed cells in body fluid, the micro-particles in water, or the microorganism in the fluid-like culture medium. The standing wave is extensively utilized as a patterning tool, and micro-particles assemble at the nodes or antinodes. For the frequencies above megahertz, the distances among nodes or antinodes are below millimeters, too close for cluster separation or detection. We create a scenario where traveling waves dominate instead of standing waves by way of scattering dissipation, and particles assemble at a central location. We first provide a theoretical prediction model based on the translational addition theorem and the partial-wave expansion method. We evaluate the attenuation exerted by an external plane wave on a set of particles immersed in the host-fluid medium, and the bulk acoustic wave (BAW) device is used to verify the prediction. It is found that, for the high concentration medium with particles, the scattering dissipation is significant and the forces are directed towards a central space instead of the nodes or antinodes of the standing waves, leading to meaningful separation of micro-particles from their host-fluid medium.

## Session 1aPPa

# Psychological and Physiological Acoustics: Spectro-Temporal Processing and Music Perception (Poster Session)

Authors will be at their posters from 9:30 a.m. to 10:15 a.m.

### *Contributed Papers*

**1aPPa1. Discrimination of rippled signals with different ripple densities.** Alexander Supin (Inctitute of Ecology and Evolution, 33 Leninsky Prospect, Moscow 119071, Russian Federation, alex_supin@mail.ru), Olga N. Milekhina, Marina S. Tomozova, and Dmitry I. Nechaev (Institute of Ecology and Evolution, Moscow, Russian Federation)

Signals with rippled frequency spectra are used to investigate capabilities of hearing to analyze and discriminate complex sounds. Till now, rippled signals with the difference between discriminated signals equal across the frequency band were exploited. In the present study, signals of different ripple densities in which ripple differences varied across the signal band were used. Ripple-density difference (RDD) thresholds increased with increasing the standard ripple density. At a standard ripple density of 2 to 10 ripples/oct, RDD threshold dependence on ripple density was identical for signal frequencies of 1 to 4 kHz: thresholds were from 0.06 ripples/oct at a standard density of 2 ripples/oct to 5–7 ripples/oct at a standard ripple density of 10 ripples/oct; RDD thresholds were not measurable at standard ripple densities above 10 ripples/oct and frequencies of 1 and 2 kHz. However, at a frequency of 4 kHz, RDD thresholds were measurable at standard ripple densities of 15 ripples/oct and higher. Hypothetically, at ripple densities of up to 10 ripples/oct, the signals were discriminated by the excitation-pattern mechanism; at ripple densities above 10 ripples/oct and a signal center frequency of 4 kHz, the signals were discriminated by the temporal-processing mechanism. [Work supported by Russian Science Foundation, Grant 16-15-10046.]

**1aPPa2. Modeling the contribution of auditory scene analysis principles to perceptual effects of orchestration.** Aurélien Antoine (Music Percept. and Cognition Lab. - Schulich School of Music, McGill Univ., 555 Rue Sherbrooke Ouest, Montreal, QC H3A 1E3, Canada, aurelien.antoine@mcgill.ca), Philippe Depalle, and Stephen McAdams (McGill Univ., Montreal, QC, Canada)

Auditory Scene Analysis (ASA) research has provided knowledge about the principles underlying auditory organization processes and has been successfully applied in computational ASA. Based on these findings, we applied these principles within a musical context. The aim is to understand and computationally model the perception of effects, such as blend and segregation, created by the combining and contrasting of properties of traditional Western instruments which result from three auditory processes: concurrent, sequential, and segmental grouping. The initial aim was to evaluate the extent to which the symbolic data provided in a musical score provide sufficient data to model the perception of these orchestral effects. Preliminary implementations have achieved an average accuracy score of 81%, suggesting that perceptual effects of orchestration can be partially retrieved by calculations based on ASA principles using symbolic data. However, many cases indicate that including properties of the acoustic signal would enhance the predictive power. This approach also provides us with the means to investigate the relative weights of the different principles involved in these grouping processes in order to understand their relative importance in musical contexts. These findings contribute to the creation of a framework for studying and understanding the perceptual characteristics of orchestration practice.

**1aPPa3. Perception of melodies and triads at high frequencies.** Daniel R. Guest (Univ. of Minnesota, 75 E River Rd., Minneapolis, MN 55455, guest121@umn.edu) and Andrew J. Oxenham (Univ. of Minnesota, Minneapolis, MN)

Accurate pitch perception is possible for harmonic complex tones with fundamental frequencies (F0s) in the musical range ($<$4 kHz) even with all harmonics beyond the putative limits of phase locking ($>$6 kHz). However, it is unknown whether this basic pitch perception supports multiple simultaneous pitch perception. To address this, we measured (1) melody discrimination with and without a complex-tone masker and (2) major-minor discrimination for triads and arpeggios composed of complex tones with low ($\sim$280 Hz) or high ($\sim$1400 Hz) F0s. The tones were filtered to ensure that in high-F0 conditions only harmonics beyond the limits of phase locking were audible. Melody perception was poorer for isolated high-F0 tones than for isolated low-F0 tones, although performance was above chance in both cases. Adding a complex-tone masker in the same spectral region degraded performance for low- and high-F0 tones. Listeners could discriminate major and minor triads and arpeggios for low-F0 tones. For high-F0 tones, some listeners could discriminate major and minor arpeggios but none could discriminate major and minor triads. These results will help elucidate whether different mechanisms underlie the perception of combinations of complex tones at low and high frequencies. [Work supported by NIH R01DC005216 and NSF NRT-UtB1734815.]

**1aPPa4. Neural encoding and perceptual decision weights associated with attention to different acoustic spectral features.** Sittiprapa Isarangura (Commun. Sci. and Disord., Univ. of South Florida, 4202 E. Fowler Ave. PCD1017, Tampa, FL 33620, sisarangura@mail.usf.edu), David A. Eddins, Erol J. Ozmeral, Robert A. Lutfi, and Ann C. Eddins (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL)

Two spectral cues that often co-vary are the spectral envelope, indexed here by spectral modulation (SM) frequency, and the modulation carrier frequency (CF). The perceptual weighting of such different spectral cues during behavioral tasks and the encoding of those weighted cues by the human auditory system is unexplored. Fifteen young normal-hearing listeners participated in two experiments. Stimuli included noise bands (three octaves wide) centered at 1500 or 1700 Hz and SM frequencies (SMF) of 2 or 4 cycles/octave. Behavioral and electroencephalographic (EEG) responses were measured simultaneously during 2-alternative, single-interval tasks. To evaluate attention to SMF versus CF cues, responses were measured in 4 conditions: passive, undirected attention, or directed attention to SMF or CF. Cortical response components were differentially modulated in attention versus passive conditions and by selective attention to SMF or CF cues. To differentiate encoding during selective attention to SMF or CF cues, perceptual decision weights were measured using a condition-on-a-single-stimulus (COSS) analysis with simultaneously recorded EEG. Results revealed individual differences in perceptual weights across listeners but distinctive cortical responses for those assigning greater weight to SMF versus CF cues. Weight- and cue-dependent neural activity indexed by source analysis will be discussed. [NIH P01AG009524; NIH DC015051.]

**1aPPa5. Benefit of tonal context on relative pitch perception in musicians and non-musicians.** Sara M. Madsen (Dept. of Psych., Univ. of Minnesota, N640 Elliott Hall, 75 East River Parkway, Minneapolis, MN 55454, madse399@umn.edu) and Andrew J. Oxenham (Dept. of Psych., Univ. of Minnesota, Minneapolis, MN)

The ability to access pitch differences between successive notes is essential for recognition of melodies and enables the detection of wrong tones in a familiar melody. The ability to compare pitch interval size is generally poor when the tones are presented without context but might improve if embedded in a tonal context. The present study measured pitch interval discrimination and aimed to determine (1) the effect of tonal context (novel or familiar melody), (2) the effect of standard interval size (2 and 9 semitones) and (3) the effect of musical training including ear training. Musicians and non-musicians between 18 and 30 years were tested. The musicians had played/sung for at least 5 years, had at least 1 semester of ear training, and still played/sung. The non-musicians had played for maximum 5 years, never had ear training lessons, and did not currently play/sing. The preliminary results show a benefit of tonal context for context presented before but not after the target interval. The benefit of the musicians is similar for the familiar and novel melodies whereas the performance of the non-musicians is markedly better for familiar compared to novel melodies. Performance is generally better for the smaller standard interval.

**1aPPa6. Investigating the parameters of temporal integration in pitch.** Anahita H. Mehta (Univ. of Minnesota, N640 Elliott Hall, 75 East River Rd., Minneapolis, MN 55455, mehta@umn.edu) and Andrew J. Oxenham (Univ. of Minnesota, Minneapolis, MN)

The pitch of the fundamental frequency (F0) can be elicited by presenting a sequence of short consecutive individual harmonics embedded in a background of noise. We conducted a series of experiments to determine the spectro-temporal properties under which this virtual pitch could be perceived. We also investigated the factors that influence the length of the maximum time window of integration for virtual pitch. F0 difference limens (F0DLs) for sequentially presented components in noise were measured in 15 normal-hearing participants. The conditions measured included harmonic and inharmonic, as well as components that would normally be spectrally resolved or unresolved when presented simultaneously. Temporal parameters included inter-tone interval (ITI) and tone duration. F0DLs in the various sequential conditions were compared with F0DLs using complex tones with simultaneously presented components. The results suggest that the percept is absent when only high-numbered harmonics ($>10$) that are resolved due to being sequential are presented as well as when the tones are inharmonically related. Results also suggest that the length of the window of integration does not differ across manipulations of ITI or tone length for resolved harmonic complexes. These results provide a unique understanding of temporal integration and resolvability in pitch through this illusory percept. [Work supported by NIH grant K99DC017472 (AHM) and R01DC005216 (AJO).]

**1aPPa7. Pitch shifts and peak selection in autocorrelation models of pitch perception.** David A. Dahlbom (Acoust., Rensselaer Polytechnic Inst., 96 11th St., Fl. 1, Troy, NY 12180, dahlbd@rpi.edu) and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

The standard autocorrelation (AC) model of pitch perception is capable of predicting a wide range of pitch phenomena, but there are a number of pitch-shift effects that it fails to predict. One of these is the pitch shift associated with a single mistuned harmonic, another noise edge pitch (NEP). A recent extension of the model by Hartmann *et al*. [*J. Acoust. Soc. Am.* **145**, 1993–2008 (2019)] employs a method that considers multiple peaks in the AC function, enabling the prediction of NEP. This extension, however, reduces the ability of the model to predict mistuned harmonic pitch. It can be shown that an additional extension to the AC model, which modifies the peak selection process, is capable of predicting both types of pitch shift. Specifically, a delay time is assigned to a peak of the AC function in a manner that takes into account the entire shape of the bump surrounding a local maximum. The end result is closely related to a template fitting procedure in the frequency domain, but it is suggested that such a procedure could correspond to processes in the time domain. The neuronal dynamics that support synchrony enhancement are suggested as a possible mechanism.

**1aPPa8. Spectral integration in temporal gap detection extends to maskers too.** David A. Eddins (Auditory & Speech Sci. Lab., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, deddins@usf.edu), Erol J. Ozmeral (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL), Eric C. Hoover (Dept. of Hearing and Speech Sci., Univ. of Maryland, Columbia, MD), Katherine N. Palandrani (Dept. of Hearing and Speech Sci., Univ. of Maryland, College Park, MD), and Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR)

Temporal gap detection is associated with obligatory and nearly perfect spectral integration. The impact of competing background noise was to determine whether such integration also applies to simultaneously presented narrowband maskers. Gap markers were 0.5 octaves wide, centered at 2000 Hz, and presented at 85 dB SPL for 400 ms. In experiment 1, marker-to-masker ratios were infinite, 12, 9, or 6 dB, gap depth was fixed at 1.0 (typical) or 0.5 (to mitigate potential masker impacts on gap depth). Thresholds (ms) were longer for the 0.5 than the 1.0 depth and systematically decreased with increasing marker-to-masker ratio. In experiment 2, gap duration was fixed (30 ms) and gap depth was the independent variable. Thresholds (depth) improved with marker-to-masker ratio as in experiment 1, though the inter-subject variability was greater. Experiment 3 included a fixed marker-to-masker ratio (12 dB) and gap depth (0.75) while masker bandwidth varied from 0.75 to 4.0 octaves. Remarkably, thresholds (ms) increased markedly for masker bandwidths from 1.0 to 4.0 octaves. Results of experiments 1 and 2 were well predicted by a temporal modulation filterbank model. To capture the masker bandwidth effect of experiment 3, model modifications included summation across modulation filters. [Work supported by NIH R01DC015051; P01AG009524.]

**1aPPa9. The perceptual simultaneity range for complex tones.** Satoshi Okazaki (Kyoto City Univ. of Arts, 13-6 Kutsukake-cho, Oe, Nishikyo-ku, Kyoto 610-1197, Japan, sat.okazaki@kcua.ac.jp) and Minoru Tsuzaki (Kyoto City Univ. of Arts, Kyoto, Japan)

Recently, we found that the range of perceptual simultaneity, within which two asynchronous pure tones are perceived to start simultaneously, shows a V-shaped curve as a function of frequency separation with the breakpoint at 0.5 Bark. This study aimed to test whether such a law of perceptual simultaneity range is applicable to complex tones. The two-tone harmonic complexes with different fundamental frequencies were used to measure the perceptual simultaneity range. Listeners were asked to judge whether the two complex tones start simultaneously or asynchronously. The use of the perceptual fusion cue for the simultaneity judgment was avoided. Results showed that the perceptual simultaneity range for two-tone complexes did not follow the above law as a function of the fundamental frequency. However, it seems that the individual components of the complex tones followed the law. These results suggest that the behavior of the perceptual simultaneity range for complex tones could be predicted by the law of perceptual simultaneity range for pure tones with considering the contributions of pure-tone components.

# Session 1aPPb

## Psychological and Physiological Acoustics: Speech Perception (Poster Session)

Authors will be at their posters from 10:15 a.m. to 11:00 a.m.

### *Contributed Papers*

**1aPPb1. Word recognition in noise and extended high frequency auditory thresholds.** Vishakha Rawool (Commun. Sci. & Disord., Univ. of MS, 1006 Briarwood Dr., Oxford, MS 38655, vishakharawool8@gmail.com) and Chelsea Campbell (Allegany Hearing & Balance, Farmington, PA)

Thirty-one individuals within the age-range of 18 to 27 years and 30 individuals within the age-range of 45 to 59 years participated in this study. Auditory thresholds were established in a sound treated booth from 250 to 16000 Hz. Words in noise scores were obtained under Sennheiser HD201 headphones through an iPad (NIH toolbox). The auditory thresholds were averaged across three frequency groups: 0.25, 0.5 and 1 kHz; 2, 3, 4, 6, 8, 9 and 10 kHz and 11.2, 12.5, 14 and 16 kHz. WIN scores were significantly correlated with all average thresholds. The correlations between the average auditory thresholds at 0.25, 0.5 and 1 kHz and WIN scores were −0.457 and −0.413 for the right and left ears. The correlations between the average auditory thresholds at 11.2, 12.5, 14 and 16 kHz were -0.699 for the right and -0.627 for the left ear. The correlation between average auditory thresholds at 2, 3, 4, 6, 8, 9 and 10 kHz for the right and left ears were −0.78 and −0.74. These findings suggest the importance of higher frequencies. [Funded by Grace Clements Communication Sciences & Disorders Research Endowment Award at West Virginia University.]

**1aPPb2. Effects of silent interstimulus interval duration between precursor and target stimuli on spectral context effects in vowel and consonant categorization.** Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu) and Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

When spectra differ between earlier (precursor) and later (target) sounds, listeners perceive larger spectral changes than are physically present. For example, when precursor sounds (e.g., a sentence) possess relatively higher frequencies, the target sound (e.g., a vowel sound in the final word) is perceived as possessing relatively lower frequencies, and vice versa. These spectral contrast effects (SCEs) have been widely reported to influence auditory perception most broadly. Broadbent and Ladefoged (1960) reported that SCEs influenced vowel categorization across varying silent interstimulus interval (ISI) durations between precursor and target sounds, with effect magnitudes appearing to decrease as ISI duration increased. Here we extended this investigation as listeners categorized vowels (/ɪ/-/ɛ/ as in "bit" and "bet") or consonants (/d/-/g/) following a context sentence with relevant frequencies amplified to produce SCEs. Precursor filter gain was + 20 dB or + 5 dB to produce larger or smaller SCEs, respectively. ISI durations were logarithmically spaced (50, 150, 450, 1350 ms). SCE magnitudes generally decreased as ISI increased. Larger SCEs decreased more quickly than smaller SCEs at longer ISIs, providing further specification to Broadbent and Ladefoged's findings. Results are consistent with the mechanisms thought to underlie SCEs, neural adaptation in both the peripheral and central auditory system.

**1aPPb3. Parameterizing effects of precursor spectral characteristics on categorization of subsequent vowel sounds.** Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu)

When spectra differ between earlier (context) and later (target) sounds, listeners perceive larger spectral changes than are physically present. When context sounds (e.g., a sentence) possess relatively higher frequencies, the target sound (e.g., a vowel sound) is perceived as possessing relatively lower frequencies, and vice versa. These spectral contrast effects (SCEs) are pervasive in auditory perception, but studies traditionally employed context stimuli with high spectrotemporal variability, obscuring precisely when spectral properties of the context biased perception. Here, contexts were two-second speech-shaped noise segments divided into four consecutive 500-ms epochs. Contexts were filtered to amplify low-F1 (100–400 Hz) or high-F1 (550–850 Hz) frequencies to encourage perception of /ɛ/ ("bet") or /ɪ/ ("bit") respectively via SCEs. Spectral peaks in the context occurred in the first 1/2/3/all 4 epochs (Onset condition), the last 1/2/3/all 4 epochs (offset condition), or only one epoch (single condition). SCE magnitudes increased as spectral peaks occurred later in the context (closer to the target), with Onset and Offset conditions producing similar results. Even brief spectral peaks in earlier epochs biased categorization. Results will be discussed with regard to pedestals, as earlier spectral peaks followed by sustained spectrally neutral context still biased perception of the subsequent vowel target.

**1aPPb4. Auditory distraction, capacity limitations, and target-distractor relationships under perceptual and cognitive loads.** Erin E. Lynch (Commun. Sci. and Disord., Ohio Univ., 53 Richland Ave., Grover Ctr., Athens, OH 45701, EL118814@ohio.edu) and Jeffrey J. DiGiovanni (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH)

Auditory distraction occurs when unwanted sounds interfere with the perception or processing of target information. Cross-modal paradigms have identified target-distractor similarities, capacity limitations, and type of task load as factors that influence distractibility. Given the scarceness of data specific to the auditory domain, this study investigated contributions from these factors across four listening tasks varying in perceptual (PL) and cognitive load (CL). Thirty adults ($M = 23.65$ years) participated in all conditions and were divided into high- or low-working memory capacity (WMC) groups following a listening-span task. Targets were spoken-digits (1-through-9) and distractors were either "Standards" (80% of trials) or one of three "Deviants" (noise-bursts, consonants, spoken-digits; 20% of trials). Experiment 1, high-PL, and Experiment 4, high-CL, were the only two conditions yielding effects of WMC and distractors. Under high-PL, high-WMC group reaction times (RTs) were both longer overall ($p = 0.026$) and in the presence of spoken-digit distractors ($p = 0.014$). In contrast, under high-CL, low-WMC group RTs were longer overall ($p < 0.01$) and spoken-digit distractors affected both groups ($p < 0.01$). Experiments 2 and 3 shared low-PL and CL, resulting in no effects. Patterns suggest that load type for a given listening task drives how capacity limitations and target-distractor similarities impact auditory distractibility.

**1aPPb5. Measuring spatial processing abilities based on attention modulating tasks.** Nirmal Kumar Srinivasan (Dept. of Speech-Lang. Pathol. and Audiol., Towson Univ., 8000 York Rd., Towson, MD 21252, nsrinivasan@towson.edu) and Lauren Charney (Dept. of Speech-Lang. Pathol. and Audiol., Towson Univ., Towson, MD)

Improvement in speech intelligibility can occur when the target talker is spatially separated from competing talkers. Reduction in an individual's hearing ability and working memory (WM) capacity are hypothesized to be two of the major reasons for reduction in performance in a complex listening environment. This study estimated working memory capacity based on a divided attention version of the classic spatial release from masking task using the Coordinate Response Measure (CRM) as speech stimuli. WM capacity was estimated with a speech source presented directly ahead of the listener and two additional speech sources either colocated at 0 deg azimuth angle or symmetrically separated by 30 deg with various amounts of temporal overlap between the speech sources. The listeners were instructed to identify all three color/number combinations presented. Adaptive procedures were used to obtain the maximum amount of temporal overlap at which listeners were still able to correctly identify the speech source presented directly ahead of the listener 50% of the time. Initial analyses of the data indicated strong relationship between SRM and temporal overlap thresholds indicating that SRM is driven by the listeners' ability to modulate their attentional mechanisms.

**1aPPb6. Spatial release from masking and working memory task.** Lauren Charney (Dept. of Speech-Lang. Pathol. and Audiol., Towson Univ., 8000 York Rd., Towson, MD 21252, lcharn2@students.towson.edu) and Nirmal Kumar Srinivasan (Dept. of Speech-Lang. Pathol. and Audiol., Towson Univ., Towson, MD)

While listening to a target talker in the presence of masking talkers, the overlap of sound energy at the cochlea (energetic masking) and confusions among speech sounds (informational masking) can reduce speech intelligibility. Improvement in speech intelligibility can occur when the target speech is spatially separated from competing speech, thus encouraging spatial release from masking. Spatial separation between target and maskers facilitate selective attention, a mechanism that depends on the availability of Working Memory (WM) capacity. Here, we present data describing the relationship between spatial release from masking and performance on attentional and working memory tasks. Working memory performance was calculated using both auditory and visual stimuli in a single-modality selective attention task and in a dual-modality selective and divided attention task. Coordinate Response Measure (CRM) sentences were used to quantify spatial release from masking. Initial analyses of the data revealed that individuals with better WM capacity had higher amounts of release from masking. These results suggest that the ability to allocate attentional resources in complex listening environments can help individuals to better understand speech.

**1aPPb7. Short-term audibility is a better predictor of vocoded speech-in-speech recognition than long-term target-to-masker ratio.** Adam K. Bosen (Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, adam.bosen@boystown.org) and Emily Buss (Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

Under natural listening conditions, speech is often heard in competing speech. In this case, the audibility of the target speech depends on short-term level fluctuations in both the target and the masker. In this experiment, we tested whether short-term audibility was a better predictor of speech-in-speech recognition than the average long-term target-to-masker ratio. 21 young adults with normal hearing listened to frozen combinations of PRESTO sentences and two-talker masking speech presented at average target-to-masker ratios between +10 and 0 dB in 2 dB steps. Frozen stimuli were vocoded with a 16-channel sine wave vocoder to discretize temporal envelope interactions between target and masker across frequencies. For each frozen stimulus we calculated short-term audibility as the proportion of time the envelope of the target speech was greater than the envelope of the masker within and across vocoder channels. Even though long-term target-to-masker ratios were set to a fixed value for each frozen stimulus, short-term audibility varied substantially across stimuli with the same target-to-masker ratio. Short-term audibility was also better predictor of speech recognition for each combination than long-term target-to-masker ratio. We conclude that short-term audibility quantifies interactions in level between fluctuating targets and maskers, whereas long-term target-to-masker ratio does not.

**1aPPb8. Comparing word and emotion recognition by listeners with normal hearing using unprocessed and vocoded speech stimuli.** Shae D. Morgan (Otolaryngol., Univ. of Louisville, 627 S. Preston St., Ste. 220, Louisville, KY 40202, shae.morgan@louisville.edu)

Current cochlear implant processing strategies preserve acoustic information necessary for speech recognition (e.g., the amplitude envelope), but significantly degrade spectral information in a signal. Listeners utilize changes in spectral information to form social judgments about a talker, such as determining the talker's sex or emotional state. Individuals with cochlear implants may be disproportionately disadvantaged when identifying this social speech information using auditory cues alone, given the device's processing emphasis on preserving word recognition only. This presentation compares the recognition of words and emotions in speech using unprocessed speech as well as speech processed using an 8-channel noise vocoder to simulate listening with a cochlear implant. We discuss the comparison of word recognition with emotion recognition for unprocessed and processed conditions across several signal-to-noise ratios (SNRs). Results indicate that emotion recognition is better than word recognition at similar SNRs, and that vocoded word and emotion recognition is poorer than when stimuli are unprocessed. The data suggest that emotion recognition is disproportionately and negatively affected by CI processing compared to word recognition.

**1aPPb9. Effects of steady background noise on benefits from voice pitch differences in a "Cocktail Party" environment.** Yonghee Oh (Speech, Lang., and Hearing Sci., Univ. of Florida, 1225 Ctr. Dr., Rm. 2128, Gainesville, FL 32610, yoh@phhp.ufl.edu), Beatrice David, Lauren Husney, and Sabrina Lee (Speech, Lang., and Hearing Sci., Univ. of Florida, Gainesville, FL)

Voice pitch is an important cue for auditory object segregation in a multi-talker environment. A recent study showed that the ability to segregate and identify speech based on voice pitch differences is highly correlated with the breadth of binaural pitch fusion, the perceptual integration of dichotic stimuli that evoke different pitches across ears [Oh *et al.*, *ASA* (2018)]. The goal of this study was to investigate effects of steady background noise on binaural pitch fusion and speech perception abilities in background noise consisting of multiple talkers. Subjects were tested in both quiet and various "soft" audible noise levels (15, 5, and -5 dB SNR) in both ears with the following experiments. First, binaural pitch fusion measurements were conducted using pure tones. Second, speech on speech masking performance was measured as the threshold target-to-masker ratio needed to understand a target talker in the presence of either same- or different-gender masker talkers. The results showed that both breadth of binaural pitch fusion was minimized and voice pitch difference benefit in multi-talker conditions was maximized at an optimal amount of the noise levels, which is called suprathreshold stochastic resonance effects. A strong negative correlation was also observed between voice gender benefit and breadth of binaural fusion. The findings suggest potential rehabilitation approaches, such as low-level noise, to sharpen binaural pitch fusion and improve related speech perception ability in multi-talker listening environments.

**1aPPb10. Importance of temporal cues in audiovisual integration in speech perception in noise.** Yi Yuan (Speech, Lang., and Hearing Sci., Univ. of Florida, 1600 SW Archer Rd., Rm. D2-77, Gainesville, FL 32610, yiyuan56@ufl.edu) and Yonghee Oh (Speech, Lang., and Hearing Sci., Univ. of Florida, Gainesville, FL)

There is a long-standing debate in the field of speech perception over the nature of the speech information that is available in the visual input. The overall goal of this project was to establish which visual characteristics can facilitate speech perception in noise, especially in conditions of competing talkers. The central hypothesis is that visual information correlated with the

temporal fluctuation of the auditory signal provides audiovisual benefits in speech perception. Unlike previous studies on audiovisual integration where articulatory gestures (i.e., facial and lip movements) were provided as the main source of perceiving speech information, a visual analog of the acoustic amplitude envelope was provided as complementary visual information to degraded auditory signals and manipulated in a series of speech perception experiments in multi-talker babble noises. The first set of experiments investigated whether speech perception in noise can be enhanced only by providing information about the amplitude envelope through a visual analog in various audiovisual coherence conditions: congruent and incongruent. The second set of experiments examined the effects of speech envelope modulation rates on audiovisual speech perception in noise. These results indicate that an abstract visual analog of acoustic amplitude envelopes can be efficiently delivered by the visual system and integrated online with auditory signals to enhance speech perception in noise, independent of particular articulation movements.

**1aPPb11. The addition of /l/ to the waveform model of vowels.** Michael Stokes (Waveform Commun., LLC, 3929 Graceland Ave., Indianapolis, IN 46208, mstokes@waveformcommunication.com)

Since its introduction, the Waveform Model of Vowel Perception and Production (Stokes, 2009) has achieved human perceptual performance on the data from two of the most cited publications in the literature (Stokes, 2011, 2014). The algorithm achieving human performance dramatically supports the Waveform Model (WM) logic, but there are phonemes not included in the Peterson and Barney (1952) data. Because of the familiarity, the Peterson and Barney (1952) data are used to illustrate the WM categories, but the WM must be able to accommodate phonemes such as /l/ to continue being a successful cognitive model. There are acoustic measurements reported for /l/, and these measurements fit perfectly into category 3 of the WM making it a natural categorical partner to /r/. Importantly, F3 measurements for /l/ are much higher than surrounding categorical vowels making it easily distinguishable from its neighbors and category 3 partner. This presentation will focus on /l/, but other phonemes appear to fit comfortably within the WM categorical pair framework as well. An algorithm is objective, and a perfect demonstration and defense of logic. Incorporating additional phonemes strengthens and extends the WM logic and programming.

MONDAY MORNING, 7 DECEMBER 2020 — 11:15 A.M. TO 12:00 NOON

## Session 1aPPc

## Psychological and Physiological Acoustics: Physiology (Poster Session)

Authors will be at their posters from 11:15 a.m. to 12:00 noon

### *Contributed Papers*

**1aPPc1. Effects of age and tinnitus: Limited physiological evidence and no perceptual consequence of cochlear synaptopathy.** Chhayakant Patro (Psych., Univ. of Minnesota, N640 Elliott Hall, 75 East River Parkway, Minneapolis, MN-5, Minneapolis, MN 55455, cpatro@umn.edu), Heather Kreft, and Magdalena Wojtczak (Psych., Univ. of Minnesota, Minneapolis, MN)

Evidence from animal models and human temporal bones has shown that cochlear synaptopathy is associated with aging. Previous work from our lab has suggested that a weakened middle-ear-muscle-reflex (MEMR) in individuals with tinnitus and normal hearing may reflect the presence of cochlear synaptopathy. In this study, we investigated effects of age and tinnitus on the MEMR strength. We also investigated the relationship between the MEMR strength and amplitudes of auditory brainstem responses (ABRs) as well as several perceptual measures that rely on precise temporal neural coding, such as amplitude-modulation detection, the detection of envelope interaural phase difference, and spatial release from speech-on-speech masking. Younger (20–30 yrs) and older (55–69 yrs) listeners with and without tinnitus and with normal or near-normal hearing were tested. We found that aging and tinnitus were associated with significant reductions in MEMR strength and ABR-based measures. However, only the MEMR-strength reduction was significant after accounting for group differences in high-frequency hearing sensitivity. No significant group differences between the older, tinnitus and control groups were found in the perceptual measures obtained in this study. Results will be discussed in terms of sensitivity of these measures to cochlear synaptopathy in humans. [Work supported by NIH grant R01 DC015987.]

**1aPPc2. Neural and behavioral correlates of change detection following training with complex auditory scenes.** Natalie J. Ball (Air Force Res. Lab., 373H Park Hall, University at Buffalo, Buffalo, NY 14260, njball@buffalo.edu), Matthew Wisniewski (Kansas State Univ., Manhattan, KS), Brian D. Simpson (Air Force Res. Lab., Wright-Patterson AFB, OH), and Eduardo Mercado (Univ. at Buffalo, SUNY, Buffalo, NY)

The effects of training on an auditory flicker task were examined in this study, where auditory scenes alternated until a participant responded "change" or "same." Change scenes differed in the location (on the horizontal plane) of two or more sounds. Half of participants were trained with auditory scenes on Day 1, and half were trained on a visual task. On Day 2, all participants were tested on auditory scenes containing either trained sounds or novel sounds within scenes; EEG was collected during testing. Participants trained with auditory scenes performed better overall than control participants. Both groups had lower reaction times to correct response change trials than same trials. Trained participants performed no better on trained sounds than novel sounds, but did perform better on trained sounds than control participants. Electrophysiologically, there were differences in ERP components based on the type of trial (change versus same auditory

scene). For same trials, N1 and P2 amplitudes were significantly higher, and P3b amplitudes were significantly lower than change trials. Additionally, P2 showed a general decrease in amplitude as scene presentations neared the response. These data show that training reduces change deafness, and improvements are not limited to the sounds experienced during training.

**1aPPc3. Steady state pressure distribution in a simple model of the cochlea with a rigid basilar membrane.** Henk Knoll (Netherlands Defence Acad., Den Helder, The Netherlands) and Arthur Vermeulen (Netherlands Defence Acad., Het Nieuwe Diep 8, Den Helder 1781 AC, The Netherlands, af.vermeulen@gmail.com)

Sound propagating through the scalae of the cochlea causes pressure changes which set the basilar membrane in motion. These pressure changes are studied in the present work with a simple model consisting of a tube with a varying diameter and a rigid wall. Analytical expressions are derived for the pressure changes in relation to the geometry of the tube, fluid friction, the acceleration of the liquid in the tube (perilymph) and stiffness of the membrane of the round window. Using a realistic parameter set, the pressure changes turn out to be dominated by the acceleration of the perilymph. The pressure changes are in phase with the acoustic stimulation and their amplitude depends on the geometry of the tube; tapering towards the middle (the helicotreme) turns out to be advantageous; the amplitudes of the differential pressure and the pressure change over the basilar membrane are maximized along the cochlear duct. Finally, the stiffness of the membrane of the round window only has impact on the static pressure in the duct.

**1aPPc4. Development of a non-parametric probabilistic model of the human middle ear.** Lucas C. Lobato (Federal Univ. of Santa Catarina, R. Delfino Conti, Florianópolis 88040-900, Brazil, lucascostalobato@gmail.com), Stephan Paul, and Júlio A. Cordioli (Federal Univ. of Santa Catarina, Florianópolis, Santa Catarina, Brazil)

The middle ear is the part of peripheral auditory system that works to transmit the sound energy from the outer to the inner ear, matching their highly different impedances. Due to its importance for the human hearing, the middle ear has been studied through mathematical models over the past 70 years. The use of deterministic approaches has been the mainstream on the development of these models despite it is well recognized that middle ear has a natural variability with respect to its mechanical and dynamical properties, kwon as random uncertainties. In this work a lumped-element model of the middle ear is used as a baseline deterministic model for the development of a probabilistic model using a non-parametric approach. This probabilistic approach is characterized by adding the uncertainties directly into the global matrices of the modeled system. Furthermore, an optimization process using a single objective function was used for fitting both probabilistic and baseline deterministic models. The probabilistic model developed presents promising results, showing statistical responses in agreement with experimental evidences. In addition, advantages were identified in comparison to a model previously developed with a parametric probabilistic approach.

**1aPPc5. Identification of listener identity from frequency following responses in 7 and 11-month-old infants.** Mark Pettet (Inst. for Learning and Brain Sci., Univ. of Washington, Seattle, WA 98195, pettetmw@gmail.com), Fernando Llanos (Dept. of Linguist, Univ. of Texas, Austin, Pittsburgh, PA), Patricia K. Kuhl (Inst. for Learning and Brain Sci., Univ. of Washington, Seattle, WA), Bharath Chandrasekaran (Dept. of Commun. Sci. and Disord., Univ. of Pittsburgh, Pittsburgh, PA), and Tian Zhao (Inst. for Learning and Brain Sci., Univ. of Washington, Seattle, WA)

The frequency-following response (FFR) is a neurophonic potential reflecting phase-locked responses from neural ensembles along the auditory pathway. In addition to encoding sound properties, the FFR also reflects listener properties. Recent work shows that adult listener identity can be decoded from the FFRs. Interestingly, listener decoding, but not stimulus decoding is greater when language experience is more limited, suggesting that the FFR represents more biometrical properties. Here we examined the extent to which biometric information is present in infants across two ages (i.e., 7 months and 11 months) that straddle the "sensitivity period" for phonetic learning. Using HMM to decode listener identity from FFRs to a non-native Mandarin lexical tone, we show that infant identity can be decoded higher than chance at both age groups with a distinct increase from 7 months (31.6%, CI ± 1.9%) to 11 months of age (45.9%, CI ± 1.6%). Listener identity, reflected in the FFRs to non-native speech, increases even as the neural commitment to non-native speech reduces [see Zhao *et al.*, ASA virtual abstract (2020)].



**1aPPc6. The effect of level-ratio paradigm on DPOAE fine structure.** Samantha K. Scheidler (Commun. Sci. and Disord., Michigan State Univ., 404 Wilson Rd., East Lansing, MI 48824, scheidl2@msu.edu), Mikayla Norton (College of Eng., Michigan State Univ., East Lansing, MI), and Maryam Naghibolhosseini (Commun. Sci. and Disord., Michigan State Univ., East Lansing, MI)

Distortion product otoacoustic emissions (DPOAEs), sound signals recorded in the ear canal and generated inside the cochlea by presenting two primary tones (f1/L1 and f2/L2; $f2 > f1$) in the ear canal, can be used to study the health and function of the cochlea. Different primary tone level-ratio paradigms (L1:L2) can be used for obtaining DPOAEs and depending on the selected paradigm, different fine structure patterns are observed. The goal of this study is to understand how three common level-ratio paradigms affect DPOAE fine structure, generated by sweeping primary tones in frequency. The paradigms used are scissors, $L1 = 0.4L2 + 39$ dB; Boys Town "optimal," $L1 = 0.45L2 + 44$ dB; and the equal-level, $L1 = L2$. The main components of the DPOAE signals, generator and reflection components, are extracted using a least squares fit algorithm. For each level paradigm, the phases of the generator and reflection components are used to determine the frequency distance between adjacent fine structure minima and maxima. This analysis is carried out for different primary levels and all paradigms. The interaction of the primary tones inside the cochlea and their impact on the fine structure pattern for each paradigm will be discussed.

**1aPPc7. Stationary solution for distortion-products in two-dimensional cochlear model with inhomogeneities.** Vaclav Vencovsky (Dept. of Radioelectronics, Czech Tech. Univ. in Prague, Technicka 2, Prague 16627, Czechia, vaclav.vencovsky@gmail.com), Ales Vetesnik (Dept. of Nuclear Chemistry, Czech Tech. Univ. in Prague, Prague, Czechia), and Anthony W. Gummer (Section of Physiological Acoust. and Commun., Dept. of Otolaryngol., Eberhard-Karls-Univ. Tubingen, Tubingen, Germany)

Vetešník and Gummer [*J. Acoust. Soc. Am.* **131**, 3914–3934 (2012)] obtained a stationary solution for distortion products (DPs) in a two-dimensional cochlear model in which the impedance of the basilar membrane varied smoothly along its length. Their solution was used to study the dependence of the nonlinear-distortion source of distortion-product otoacoustic emission (DPOAE) on stimulus level [Vencovský *et al.*, *J. Acoust. Soc. Am.* **145**, 2909–2931 (2019); Vencovský *et al.*, *J. Acoust. Soc. Am.* **146**, EL92–EL98 (2019)]. Here, the solution is extended to "impedance irregularities" in the mechanics of the cochlea causing the generation of the coherent-reflection source of DPOAEs near the DP characteristic place. The coherent-reflection source of DPOAEs can be relatively pronounced in normal-hearing humans and the interference between the nonlinear-distortion and coherent-reflection components complicates the use of DPOAEs in routine clinical screening of the auditory periphery. The solution for the model with mechanical perturbation can be used to study the relationship between nonlinear-distortion and coherent-reflection components of DPOAEs.

**1aPPc8. Damped ultrasonic vibro-acoustic behavior of temporal bone structures.** Christopher M. Dumm (Dept. of Mech. Eng. and Mater. Sci., Univ. of Pittsburgh, 3700 O'Hara St., 636 Benedum Hall, Pittsburgh, PA 15261, cmd113@pitt.edu), Anna C. Hiers, David B. Maupin, Jeffrey S. Vipperman (Dept. of Mech. Eng. and Mater. Sci., Univ. of Pittsburgh, Pittsburgh, PA), George E. Klinzing (Dept. of Chemical and Petroleum Eng., Univ. of Pittsburgh, Pittsburgh, PA), and Carey D. Balaban (Dept. of Otolaryngol., Univ. of Pittsburgh, Pittsburgh, PA)

Little research exists describing how the human auditory and vestibular systems respond to higher intensity pressure at ultrasonic frequencies. In large part, this literature gap arises from the well-known attenuation of frequencies higher than 20 kHz by the air-conduction hearing path. Most studies of bone-conducted sound also avoid characterization of ultrasonic sound transmission, in part because the basilar membrane of the cochlea is seen as a physical Fourier analyzer which does not have sensitivity above 20 kHz. We used finite-element modeling of human and macaque intracranial structures to investigate transmission of modulated airborne ultrasound signals in the skull base, middle ear, and inner ear. Even in the presence of soft tissue damping, a family of resonant structural features could locally amplify sound within ultrasonic frequency range. High-amplitude, air-propagated ultrasound carrier signals, heterodyned with audio-band signals, may be able to use these resonant features to deliver perceptible signals to the cochlea without activation of the vestibular sensors. These principles offer the potential to develop new, less obtrusive, noncontact equipment enabling personal communication. [Work supported by ONR Grant N00014-18-1-2157.]

**1aPPc9. Neural correlates of extremely large individual differences in interaural cross-correlation detection.** Angkana Lertpoompunya (Commun. Sci. and Disord., Univ. of South Florida, 4202 E Fowler Ave., PCD 1017, Tampa, FL 33620, angkanal@mail.usf.edu), Erol J. Ozmeral, Nathan Higgins, Ann C. Eddins, and David A. Eddins (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL)

Interaural cross-correlation is a primary feature that can predict the detection of dichotic signals in background noise. Perfectly correlated noise presented to the two ears is perceived as compact and centrally lateralized. As noise is binaurally de-correlated, the perceptual image widens and becomes more diffuse. Previous investigators have reported large variability across individuals' cross correlation just-noticeable differences (IACC-JNDs). Anecdotally, many listeners with normal pure tone thresholds present as extreme outliers in binaural tasks. It is unknown whether or not this variability across individuals is reflected in the neural activity recorded during passive listening to these stimuli. IACC-JNDs were measured in a behavioral task and compared to correlation-change evoked potentials using electroencephalography (EEG). Results indicate significantly smaller cortical responses (N1 and P2 amplitudes) to interaural change differences for individuals with large- compared to small-JND thresholds. Source localization of this effect was observed in multiple subdivisions of the auditory cortex, include Heschl's gyrus, planum temporale, and the temporal sulcus. These results provide objective electrophysiological evidence of a binaural processing deficit corresponding to extremely large JND thresholds. Interestingly, these listeners are indeed binaurally disadvantaged despite otherwise normal audiometric profiles and lack of spatial hearing complaints. [Work supported by NIH P01AG009524.]

**1aPPc10. Real-time implementation of an auditory nerve model using a system-on-chip field-programmable gate array.** Matthew Blunt (Elec. and Comput. Eng., Montana State Univ., Culbertson Hall, Bozeman, MT 59717, mblunt97@yahoo.com), Hezekiah Austin, Trevor Vannoy (Elec. and Comput. Eng., Montana State Univ., Bozeman, MT), Tyler Davis (Flat Earth, Inc., Bozeman, MT), and Ross Snider (Elec. and Comput. Eng., Montana State Univ., Bozeman, MT)

System-on-Chip (SoC) Field Programmable Gate Arrays (FPGAs) are ideal for real-time signal processing due to their low, deterministic latency and high performance. We target our Audio Blade platform that contains an Intel Arria 10 SoC FPGA with floating-point capability and 1.5 TFLOPS performance. In order to target the Arria 10, the auditory nerve model [1, 2] needed to be ported to a hardware description language. We accomplished this by first porting the MATLAB/C model to Simulink, and then used MathWorks HDL Coder to generate VHDL code. Our hardware-accelerated model will allow researchers to edit model parameters in real-time from Linux software running on the embedded ARM CPUs and view the resulting nerve responses in real-time. The goal is to create a real-time platform running multiple auditory nerve models, allowing researchers to inject hearing impairments and then develop hearing aid strategies to compensate for these hearing deficits. We will discuss performance measures of the FPGA-based auditory nerve models, including latency measures and how many auditory nerve models can run simultaneously in the Arria 10 FPGA fabric. 1. Bruce *et al.*, *Hearing Res.* **360**, 40–54 (2018). 2. Zilany *et al.*, *JASA* **126**, 2390–2412 (2009).

**1aPPc11. Endogenous brain oscillations in the 10–20 Hz range during auditory spatial attention.** Jordan Love (SLHS, Purdue Univ., Lyles-Porter Hall, 715 Clinic Dr., West Lafayette, IN 47907, oliver49@purdue.edu), Barbara Shinn-Cunningham (Carnegie Mellon Neurosci. Inst., Carnegie Mellon Univ., Pittsburgh, PA), and Hari Bharadwaj (Speech, Lang., and Hearing Sci., Purdue Univ., West Lafayette, IN)

Oscillatory brain activity in the alpha frequency range (7–13 Hz) is commonly implicated in the attention literature, though the exact role it plays in facilitating attention is debated. A popular functional theory, the "alpha suppression hypothesis," suggests that changes in alpha amplitudes can be interpreted as a marker of active suppression of neural assemblies that code for distractors. Recent studies of visual selective attention have questioned this hypothesis. In the auditory attention literature, alpha power during listening has been reported to fluctuate with the temporal structure of target sounds. Yet, studies examining whether purely endogenous alpha activity during orienting of attention predicts trends in subsequent task performance are limited. The current study addresses this gap by quantifying changes in alpha power observed in the electroencephalography (EEG) signal during an auditory selective attention task. This analysis focuses on the preparatory period, a window of time following a directional attention cue but preceding the auditory stimulus. Preliminary results suggest the presence of robust endogenous oscillations between 10 and 20 Hz during this preparatory period. Further analysis will compare the strength of these rhythms to individual differences in task performance, to variations in task demands, and to within-individual trial-to-trail variations in perceptual outcomes.

## Session 1aSCa

## Speech Communication: Clinical Studies in Children's Speech (Poster Session)

Authors will be at their posters from 9:30 a.m. to 10:15 a.m.

### *Contributed Papers*

**1aSCa1. Maternal depression and the timing of mother-child dialogue.** Valerie F. McDaniel (Speech, Lang. and Hearing Sci., Univ. of Missouri, 301 Lewis Hall, Columbia, MO 65211, valerie.f.mcdaniel@health.missouri.edu), Chloe M. Sowell, Katherine A. Boley, Janelle Janssen (Speech, Lang. and Hearing Sci., Univ. of Missouri, Columbia, MO), Jean M. Ispa (Human Development and Family Sci., Univ. of Missouri, Columbia, MO), and Nicholas A. Smith (Speech, Lang. and Hearing Sci., Univ. of Missouri, Columbia, MO)

Early social interaction lays the foundation for developing language skills. Maternal depression may affect social interaction by disrupting the temporal structure of turn-taking. In this study, we examine the temporal properties of dialogue extracted from video recordings of semi-structured play between mothers and preschool children enrolled in the Early Head Start Research and Evaluation Program (n = 110). Our preliminary analyses examine how higher levels of maternal depression (on the CES-D scale) are related to the number and duration of utterances, the number of conversational turns, and the latency and variability of responses for both the mother and the child. The primary goal of the study is understanding how depression impacts bidirectional processes in mother-child interaction, and the quality of children's early language experience.

**1aSCa2. Preschool boys with hearing loss are exposed to greater vocal amplitude from parents.** Mark VanDam (Washington State Univ., 412 E. Spokane Falls Blvd, SHS, HSB 125-X, Spokane, WA 99202, mark.vandam@wsu.edu), Sean Morton, and Karen E. Brown (Washington State Univ., Spokane, WA)

Children's ability to perceive auditory signals impacts speech and language development, cognition, academic performance, and social relationships. Vocal signals in the family environment produced by mothers, fathers, and children have different characteristics and likely contribute differently to development. Childhood hearing loss has been associated with developmental progress in children in some (but not all) domains. In particular, the level of auditory input to children from specific talkers is not well documented. The aim of this study is to examine the amplitude of speech from mothers and fathers in the presence of preschool boys and girls with and without hearing loss. We processed 688 daylong audio recordings using automatic speech processing routines to identify the amplitude of the vocal utterances of mothers (n = 1 368 108 utterances) and fathers (n = 591 335 utterances) when in the presence of their toddlers with and without hearing loss. Results suggest both mothers and fathers use increased vocal amplitude with children with hearing loss but that effect is driven by increased amplitude to boys with hearing loss. This sex effect may shed light on the different roles that family members play in the socio-cultural development of speech and language use.

**1aSCa3. Effects of speech and sung speech training on speech prosody production by trilingual children with autism spectrum disorder.** Si Chen (Chinese and Bilingual Studies, The Hong Kong Polytechnic Univ., Hong Kong 00000, Hong Kong, qinxi3@gmail.com), Bei Li, Fang Zhou, Angel Wing Shan Chan, Tempo Po Yi Tang, Eunjin Chun, Phoebe Choi, Chakling Ng, Fiona Cheng, and Xinrui Gou (Chinese and Bilingual Studies, The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong)

Speech prosody can be used to distinguish old (topic) versus new (focus) information and rejecting incorrect alternative statement. It has been reported that children with autism spectrum disorder (ASD) may show abnormal prosody of either monotonic or exaggerated intonation and some of them may fail to mark focus. Though speech and musical training have shown to improve speech production by ASD children, no specific training methods have been proposed to improve the use of speech prosody to mark focus and few studies investigated tonal language speakers. We aim to test (1) whether Cantonese-speaking ASD children fail to mark focus in their native tonal language (2) whether trainings may improve the speech prosody processing (3) whether sung speech training is more effective than speech training? We recruited two training groups of Cantonese-speaking ASD children, a control group of ASD children and TD children. In the training tasks, we focused on improving the mapping between the acoustic cues and information structure. Our pilot results showed that speech and musical training improved the use of prosodic cues such as intensity and f0 in marking focus across various positions. However, ASD children may have difficulties in integrating all the prosodic cues across conditions.

**1aSCa4. Advancing speech activity detection for automatic speech assessment of pre-school children prompted speech using COMBO-SAD.** Satwik Dutta (CRSS-Ctr. for Robust Speech Systems, Univ. of Texas at Dallas, 800 West Campbell Rd., Richardson, TX 75080, satwik.dutta@utdallas.edu), Prasanna Kothalkar (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas - Dallas, Richardson, TX), Johanna Rudolph, Christine Dollaghan, Jennifer McGlothlin, Thomas Campbell (Callier Ctr. for Commun. Disord., Univ. of Texas at Dallas, Richardson, TX), and John H. Hansen (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas - Dallas, Richardson, TX)

Speech sound disorder (SSD), a developmental disorder which affects children's ability to produce the sounds (and words) within their native language, has a prevalence rate of 3%–16% among children in USA. Screening for SSDs generally requires recording, evaluation, and decision-making by a certified speech-language pathologist (SLP). Automating part or all of this process could significantly reduce the amount of time and effort in the screening process. However, in order for this process and especially the final "pass"/"fail" screening decision to be automated, children's speech content must be extracted from within a collected audio sample and therefore requires speech/silence activity detection. For this study, an iOS application for field use was developed to collect speech word productions from children, with algorithmic processing of all participants assigned a Percentage of Consonants Correct-Revised (PCC-R) score by a certified SLP. An unsupervised speech-activity-detection (SAD) algorithm is explored. COMBO-SAD, originally developed during the DARPA-RATS program, was modified to for use on child speech. Model evaluation was performed on a diverse collected child corpus based on their PCC-R score. Finally, a

duration "shoulder" extension of SAD boundary labels was also analyzed to benchmark potential system impact on model performance.

**1aSCa5. Child hearing loss and vocal turn-taking between family members.** Kaelin Kinney (Psychol. and Brain Sci., Univ. of Louisville, 2301 S 3rd St., Louisville, KY 40292, kmkinn03@louisville.edu), Maria V. Kondaurova (Psychol. & Brain Sci., Univ. of Louisville, Louisville, KY), Mark VanDam (Speech & Hearing Sci., Washington State Univ., Spokane, WA), and Qi Zheng (Biostatistics, Univ. of Louisville, Louisville, KY)

How does hearing loss affect vocal turn-taking within families? This study examined turn-taking between children and multiple members of their social environment. For children with hearing loss, we also examined potential differences by device, hearing aids (HA) versus cochlear implants (CI). Daylong audio recordings were obtained monthly for about a year using a wearable recorder. Conversational turns per hour (CTC/hr) between children with- and without hearing loss and their family members were estimated by automated speech processing. Results indicate that the CI children engaged in fewer CTC/hr with adult caregivers compared to the HA and normal-hearing groups. Initially, a higher CTC/hr between the target child and the adult female was observed compared to the adult male or the other child. With child age, turn taking between the target child and the adult female increased relative to the target child and the adult male. Over time, CTC/hr between the target child and other child exceeded turn taking between others. This increase occurred earlier in families with siblings. Results suggest vocal turn-taking between family members depends on the degree of child hearing loss and relations within the family. Longitudinally, there was a positive effect of assistive device on the quantity of turns.

**1aSCa6. Analysis of acoustic cue production for atypical speech diagnosis.** Soo Jung Jang (MIT, 77 Massachusetts Ave., Cambridge, MA 02139, soojungj@mit.edu), Jeung-Yoon Choi (MIT, Cambridge, MA), and Stefanie Shattuck-Hufnagel (MIT, Cambridge, MA)

Acoustic cues to lexical distinctive features can be found by examining speech waveform and spectrogram measurements, and can provide a more detailed analysis of speech than is currently provided by methods for identifying atypical speech production. Applying this individual-feature-cue framework to the clinical diagnosis domain, this project analyzes the acoustic cue patterns of young patients with various diagnoses, such as autism spectrum disorder, dyslexia, and Specific Language Impairment. It is hypothesized that each group will produce distinctive acoustic cue patterns for a widely used non-word repetition task, C-TOPP. The speech recordings of 37 subjects were labeled with their acoustic cues; these were then aligned with the cues expected in a canonical production, and acoustic cue level discrepancies were identified. With mispronunciation patterns of each atypical group identified, a Support Vector Machine (SVM) model was developed that can computationally help with diagnosing a patient, given his/her speech recording for this task. Although there are limitations to the study's diagnosis model due to the small data set, the preliminary results show promising insights, yielding 75.7% prediction accuracy for the diagnostic group.

**1aSCa7. Acoustic and articulatory development in deaf and heard of hearing children after pediatric auditory brainstem implantation.** Jolien Faes (Linguist, Univ. of Antwerp, Lange Winkelstraat 40-42, Antwerp 2000, Belgium, jolien.faes@uantwerpen.be) and Steven Gillis (Linguist, Univ. of Antwerp, Antwerp, Belgium)

Auditory brainstem implantation (ABI) is still in its infancy in pediatric populations (Puram and Lee, 2015). For children born with a severe-to-profound hearing loss who cannot benefit from cochlear implants (CI), ABI is shown a valid option for (partial) hearing restoration (Sennaroglu *et al.*, 2016). Research has already showed the perceptual benefits after ABI implantation (Sennaroglu *et al.*, 2016). However, fairly little is known about the effect of ABI implantation on speech production development. We aim to study the acoustic and articulatory development of these children in comparison to children with CI and typical hearing. Our design is longitudinal: the spontaneous speech of three cases is recorded monthly up to four years after implantation. Acoustic features of vowels after two years of device use suggest smaller vowel spaces in children with ABI than in children with CI. There is a clear effect on phonological development: the children produce an increasing amount of ambient language phonemes with longer device use and extended vocabulary size. Nevertheless, the children have lower phonological complexity and production (accuracy) then children with CI and typical hearing. Despite the (individual) differences, ABI implantation shows a positive effect on oral language production in children with severe-to-profound hearing loss.

**1aSCa8. Development of Indonesian audiovisual speech synthesis system for assistance children with delayed speech.** Elok Anggrayni (Eng. Phys., Institut Teknologi Sepuluh Nopember, Surabaya, Indonesia), Dhany Arifianto (Eng. Phys., Institut Teknologi Sepuluh Nopember, Sukolilo Campus, Surabaya 60111, Indonesia, dhany@ep.its.ac.id), Nyilo Purnami (Otorhinolaryngology Head and Neck Surgery, Airlangga Univ., Surabaya, Indonesia), Joko Sarwono (Eng. Phys., Bandung Inst. of Technol., Bandung, Indonesia), and Sangsaka Wira (Eng. Phys., Institut Teknologi Sepuluh Nopember, Surabaya, Indonesia)

Hearing impairment is one of the congenital deafness frequently found in children, which is followed by a delayed speech. Furthermore, a speech therapist currently available is limited. In this research, we outlined the development of the Indonesian audio-visual speech synthesis system for learning of the deaf children with delayed speech. First, we developed two kinds of Indonesian corpus, such as speech corpus and audio-visual corpus. The speech corpus contains speech recordings from professional speech therapists. The total duration of all recorded Indonesian speech database is more than 18 hours of audio. The audio-visual corpus contains visual phoneme (viseme) which is the visualization of Indonesian phoneme for lips. Segmentation and labeling were conducted to create transcriptions. We did some variation in the number of sentences and the type of sentences used in the training part of speech synthesis. Audio-visual synthesis used viseme concatenation method. The objective evaluation result using the Mel-cepstrum distortion method was 2.8. The subjective evaluation result using Mean Opinion Score was 3.71. The evaluation results showed that the new design of Indonesian audio-visual speech synthesis for learning to produce any single meaningful word was capable to use as the alternative for hospitals for the therapy of the delayed speech patients.

**1aSCa9. Classification of accurate and misarticulated rhotic syllables for simplified ultrasound biofeedback therapy.** Sarah R. Li (Biomedical Eng., Univ. of Cincinnati, 231 Albert Sabin Way, CVC, 3960, Cincinnati, OH 45267, lisr@mail.uc.edu), Sarah Dugan (Commun. Sci. and Disord., Univ. of Cincinnati, Dayton, OH), Colin Annand (Psych., Univ. of Cincinnati, Cincinnati, OH), Kathryn Eary, Michael Swearengen, Gregory A. Terrell, Sarah Stack (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), Michael A. Riley (Psych., Univ. of Cincinnati, Cincinnati, OH), and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH)

Ultrasound biofeedback therapy (UBT) provides real-time imaging of tongue movements and has demonstrated positive speech remediation outcomes; however, some individuals have limited or no response. UBT outcomes could be further improved by a simplified biofeedback display to enhance motor learning. Such simplification requires automatic processing of ultrasound images to determine biofeedback parameters and targets. We investigate potential biofeedback parameters using TonguePART, a method that automatically tracks the tongue surface on midsagittal ultrasound images to quantify displacement trajectories of the tongue root, dorsum, and blade. Our focus is rhotic syllables (/i/, /u/, /o/, /e/, /ɛ/, and /a/ with initial or final /r/) from children with residual speech sound disorders and children with typically developing speech. We train support vector machines on measured tongue part displacement trajectories to distinguish between accurate and misarticulated productions as determined from auditory perceptual ratings. Preliminary data indicate that a linear combination of the tongue dorsum and blade displacements, between the vowel and consonant, can distinguish between accurate and misarticulated productions of rhotic syllables. These results suggest a real-time biofeedback parameter based on projections of real-time dorsum and blade displacements, along with potential target values, different for each vowel, for this parameter in simplified UBT for speech remediation.

# Session 1aSCb

## Speech Communication: Speech Production I (Poster Session)

Authors will be at their posters from 10:15 a.m. to 11:00 a.m.

### *Contributed Papers*

**1aSCb1. Mapping Southern spoken dialect features with geographic information systems.** Jonathan Jones (Dept. of Geography, Univ. of Georgia, Athens, GA 30602, jonx@uga.edu) and Margaret E. Renwick (Linguist, Univ. of Georgia, Athens, GA)

We use GIS mapping to analyze spatial trends in spoken language, testing how features identified as part of the 'Southern dialect' by the *Atlas of North American English* (*ANAE*) are realized in the Digital Archive of Southern Speech (DASS). We analyze mergers, diphthongization, monophthongization, fronting, g-dropping, and rhoticity. Acoustic data from DASS was analyzed using R, generating feature-appropriate summary statistics. GIS analysis was conducted with GeoDa, QGIS, and ArcGIS Online. Spatial analysis used the Local Moran's I method to identify geographic clusters of similar values. Generally, DASS data agrees with *ANAE*'s descriptions. However, POOL-PULL are not consistently merged, unlike in *ANAE*, except for a cluster in Eastern Tennessee. /ɔɪ/-monophthongization also varies: averages indicate the vowel was still fairly diphthongized, while spatial autocorrelation finds monophthongization in Central Tennessee and Atlanta. /oʊ/ appears to front only weakly; however, the greatest fronting is found in Florida. We find high rates of g-dropping and rhoticity, but mapping and analysis for these features reveals clusters of more g-dropping in Eastern Tennessee, less rhoticity along Florida's Gulf Coast, and more rhoticity in Northern Mississippi. Our analysis partially corroborates *ANAE*'s description, suggesting that Southern speech changed between the DASS interviews and the later publication of *ANAE*.

**1aSCb2. Coarticulatory assimilation of velar consonants on vowels in the Southern Dialect of America English.** Jaden Bouguyon (Linguist, Emory Univ., Emory University 532 Kilgo Circle 202C, Atlanta, GA 30322-0001, jbouguy@emory.edu), Ryan Seitter, and Yun J. Kim (Linguist, Emory Univ., Atlanta, GA)

Previous studies have shown dialectal differences in assimilatory coarticulation patterns. For example, Southern American English (i.e., SAE) speakers have more nasalized vowels in pre-nasal contexts than non-SAE speakers (e.g., Awan *et al*., 2015). This study asks whether SAE and non-SAE speakers differ in another assimilatory coarticulation process, namely, the "velar pinch," or the convergence of F2 and F3 in pre-velar vowels. To answer this question, a production study is conducted: 5 SAE speakers and 5 non-SAE speakers are recruited to produce 24 words embedded in sentences. The target words include three types of assimilatory contexts: pre-/k,g/, pre-/ŋ/, and pre-/n/ vowels. The degree of vowel nasalization and the formant transitions before the consonant are analyzed based on Kaiser (1997). Preliminary results show that SAE speakers have stronger vowel nasalization in the pre-/n/ and pre-/ŋ/ contexts, and smaller F3-F2 differences in the pre-/k,g/ contexts, suggesting that SAE speakers show not only stronger vowel nasalization but also stronger "velar pinch" than non-SAE speakers. Interestingly, in the pre-/ŋ/ contexts in which both types of coarticulatory assimilation are possible, SAE speakers have weaker "velar pinch" than non-SAE speakers. Sociolinguistic implications of the results will be discussed.

**1aSCb3. An acoustic analysis of St. Lawrence Island Yupik vowels.** Benjamin Hunt (English, George Mason Univ., 4400 University Dr., 3E4, Fairfax, VA 22030, bhunt6@gmu.edu), Harim Kwon, and Sylvia Schreiner (English, George Mason Univ., Fairfax, VA)

St. Lawrence Island Yupik (ISO 639-3 ess; henceforth Yupik) is an endangered language spoken by 800–900 speakers in Alaska and Russia (Schwartz *et al*., 2020). Minimal research has been conducted on Yupik phonology, with its vowel inventory being impressionistically described as consisting of seven vowel phonemes: /i/, /iː/, /a/, /aː/, /u/, /uː/, and /ə/ (e.g., Krauss, 1975). We present the first known acoustic examination of Yupik vowels, using data from four native speakers. Materials included words beginning with the seven previously reported vowel phonemes followed by an obstruent varying in its place of articulation (labial, coronal, velar, uvular). Statistical analyses on duration and formant frequencies were combined with visualizations of the vowel space using normalized F1-F2. The findings largely aligned with previous descriptions of the seven vowel phonemes, with the phonemic length distinction primarily realized in duration. We furthermore found that the mid-vowels were mostly reduced (shorter in duration and often devoiced) in word-initial position; /i/, /iː/, and /e/ were backed and lowered when followed by a uvular obstruent; and non-front vowels were fronted in the coronal environment. The current findings suggest potential undocumented vowel allophonies in Yupik, including coarticulatory assimilation as well as vowel reduction.

**1aSCb4. Formant and voice quality changes as a function of age in women.** Laura Koenig (Adelphi Univ., 300 George St., New Haven, NY 06511, koenig@haskins.yale.edu), Susanne Fuchs (Leibniz-Ctr. General Linguist, Berlin, Germany), Annette Gerstenberg (Univ. of Potsdam, Potsdam, Germany), and Moriah Rastegar (Adelphi Univ., Garden City, NY)

Numerous studies have assessed effects of aging on the voice, but there remains some lack of consensus on the nature and magnitude of such effects. Although discrepancies may arise from methodological factors, well-controlled studies of aging also show substantial individual differences. Documented changes in laryngeal tissues and vocal tract dimensions suggest that aging may have various effects on the voice, and measures of aperiodicity, vowel formant frequencies, and speaking fundamental frequency have been frequently employed. Rather few studies have assessed spectral measures of voice quality, however, although changes in vocal fold thickness or glottal aperture may be expected to affect such metrics, as can individual differences in lifelong voice use and care. This study uses longitudinal data from French female speakers obtained as part of the LangAge project (langage-corpora.org). High-quality narrative samples were obtained 7–10 years apart from speakers in their 1970s at the first recording. Vowels from frequently repeated words were extracted from the samples. Along with fundamental frequency and formants, we also assess spectral tilt measures and spectral noise. By using multiple measures, we hope to gain insight into the range of ways in which speakers' voices may show effects of aging.

**1aSCb5. Non-modal sonorants in Hakha Lai.** Kelly H. Berkson (Linguist, Indiana Univ., 1020 E. Kirkwood Ave., Ballantine Hall 852, Bloomington, IN 47405-2201, kberkson@indiana.edu) and Stefon M. Flego (Linguist, Indiana Univ., Bloomington, IN)

Phonemic non-modal sonorants are both typologically rare and prone to diachronic loss crosslinguistically, but are robustly attested and diachronically stable in subgroups of Tibeto-Burman (TB) such as Kuki-Chin (KC). KC is a cluster of appr. 50 related languages spoken in Chin State in western Burma/Myanmar and by a large refugee community in Indiana. Chin languages generally contain rich phonetic inventories replete with 3-way laryngeal stop contrasts (e.g., /t $t^h$ d/), lateral affricates (/tl, $tl^h$/), and modal/non-modal sonorant pairs (most commonly /m m̥, n n̥, *η η̊*, l l̥, r r̥/). Extant phonetic investigation of non-modal sonorants—in KC and beyond—is limited, and often focuses exclusively on the nasals, but recent data suggest highly variable phonetic instantiation across languages, speakers, and contexts that may involve true voicelessness, slack or breathy voicing, and pre- or post-aspiration. The current study presents acoustic analysis of the full suite of modal and non-modal sonorants in Hakha Chin. Data are from six native speakers (three female) now living in Indiana. A variety of temporal and spectral measures are reported in order to provide a more nuanced understanding of the degree and type of variation observed in production of these sounds.

**1aSCb6. Effect of consonants on onset F0: Evidence from Kansai Japanese.** Miao Zhang (Linguist, Univ. at Buffalo, SUNY, 2213 Sweet Home Rd., Apt. 24, Buffalo, NY 14228, miaozhan@buffalo.edu)

This study investigates the effect of voicing on onset F0 in Kansai Japanese. In tone languages, the effect of initial consonants on F0 tends to be smaller and shorter, hypothetically to maximize the tonal contrast preserved for lexical items. A recent study (Gao and Arai, 2019) found that despite Tokyo Japanese being a pitch-accent language, the consonant effect on F0 was not necessarily inhibited. However, this might be due to decreased laryngeal control in Tokyo Japanese because of its simple tonal contrast. Since the pitch accent system of Kansai Japanese is more complex than that of Tokyo Japanese, Kansai Japanese should show less F0 perturbation. In the experiment, five native speakers (4 females, 1 male) produced the target words /CVma/ (C: /n, b, p/, V: /i, a/) in different pitch accent types (HH, HL, LH) and focus conditions (broad, narrow, contrast focus). My data show that overall, F0 following /p/ is significantly higher. The more complex pitch-accent system did not lessen the F0 perturbation. Consistent across focus, the effect is slightly larger in words with a high pitch accent and in the high vowel /i/ context. This may be due to the congruence in laryngeal settings for high F0 production.

**1aSCb7. Postaspiration in Sevillian Spanish: Beyond voiceless stops.** Madeline Gilbert (Linguist, New York Univ., 10 Washington Pl., New York, NY 10003, mg5171@nyu.edu)

Sevillian Spanish /s + ptk/ clusters are often produced with gemination of the consonant following /s/ and postaspiration (/pasta/:[pahta]–>[pat:$^h$a]) (Ruch and Peters, 2016). Spanish voiceless stops are unaspirated, but some argue that postaspirated stops are phonologizing in Seville (O'Neill, 2010; Gylfadottir, 2015). Acoustically, postaspiration is long VOT. Articulatorily, it is described as a laryngeal gesture moving across the stop, continuing past the release (Torreira, 2007). Previous work focuses only on /s + ptk/. This acoustic study examines /sC/-clusters with voiceless stops, voiced stops, and sonorants, extending our understanding of the mechanisms behind postaspiration. Seven Sevillians read paragraphs containing /sC/ and /C/ words (/kaspa/-/kapa/). All clusters show gemination. Compared to /ptk/, /s + ptk/ clusters have long postaspiration, resist intervocalic voicing, and result in higher pitch on the following vowel (a cue to contrastive aspiration in many languages, Dmitrieva *et al*., 2015). /s + bdg/ clusters have stronger constriction, higher COG, and less voicing than intervocalic (spirantized) /bdg/. /s + mnl/ clusters differ from /mnl/ only in gemination. Gestural overlap thus occurs in voiced and voiceless clusters, and appears to result from higher-level realignment as opposed to unintentional mistiming (Parrell, 2012). Postaspirated voiceless stops are unlikely to phonologize because the process occurs with both /s + bdg/ and /s + ptk/, giving different results.

**1aSCb8. The influence of inter-dialect contact on the Korean three-way laryngeal distinction: An acoustic comparison among Seoul, Gyeongsang, and bidialectal speakers.** Hyoju Kim (Linguist., Univ. of Kansas, 1541 Lilac Ln., Blake Hall, Rm. 427, Lawrence, KS 66045-3129, kimhj@ku.edu) and Allard Jongman (Linguist., Univ. of Kansas, Kansas City, KS)

This study investigates the acoustic correlates of the Korean three-way laryngeal stop distinction in Gyeongsang-based Seoul bidialectal speakers who were born in the Gyeongsang region but moved to Seoul to pursue higher-level education. The main purpose is to examine whether exposure to Seoul Korean affects bidialectal speakers' cue-weighting strategy to distinguish stops in production. Acoustic data were collected from 8 bidialectal, 5 Gyeongsang speakers (GK), and 11 Seoul speakers (SK). Bidialectal speakers produced stimuli in both Gyeongsang and Seoul dialect. A cue-weighting model of measured data reveals that VOT is less important to distinguish lenis from aspirated stops for SK and for bidialectal speakers' SK compared to GK and bidialectal speakers' GK. In addition, F0 is more important for the lenis-aspirated distinction for bidialectal speakers' SK and for SK compared to GK and bidialectal speakers' GK, showing that bidialectal speakers rely less on VOT and more on F0 to distinguish lenis from aspirated stops compared to GK. Bidialectal speakers' SK reveals that they rely more on VOT and less on F0 compared to SK. These results provide empirical support to previous studies (e.g., Lee & Jongman, 2019) suggesting that a series of changes in GK are due to inter-dialect contact.

**1aSCb9. Production of the vowels in the second person pronoun in Seoul Korean.** Jiseung Kim (Linguist, Univ. of Michigan, 416 S 5th Ave., Apt 4, Ann Arbor, MI 48104, jiseungk@umich.edu) and Yoonjeong Lee (Linguist, Univ. of California, Los Angeles, Los Angeles, CA)

The goal of the study is to investigate generational differences in the production of the vowels following /n/ in the second person singular pronouns /nʌ/ and /ni/ and the first person singular pronoun /ne/ in Seoul Korean. Due to the merger between the low front vowel /æ/ and the high-mid front vowel /e/, the semantic distinction between the first person pronoun and the second person pronoun was lost. Chae (1997) argued that it was the basis for the development of the new form of the second person pronoun /ni/ in Seoul Korean. However, no study as of yet has examined whether there are generational differences in the use of the new form with different particles and in different informational structure. The current study hypothesizes that there will be differences between younger and older speakers in the acoustic properties of the vowels in the second person pronoun that are modulated by the type of the postpositional particles as well as the informational structure of the utterances in Seoul Corpus (Yun *et al*., 2015).

**1aSCb10. Do children compensate for their vocal tract morphology during vowel production?** Margaret Cychosz (Dept. of Linguist, Hearing and Speech Sci., Univ. of Maryland, College Park, 1203 Dwinelle Hall #2650, Berkeley, CA 94720, mcychosz@berkeley.edu)

Children's vocal tracts (VT) are not miniature adult versions: the oral cavities are longer relative to the pharyngeal. Children may compensate for this morphology by shifting constriction location to approximate adult-like resonances (Ménard *et al*., 2007). But the same lingual adjustment may be insufficient across development because children exhibit non-uniform VT growth. The constriction location required to create a resonator at age 4 will differ from that required at age 8. For child formant normalization, then, uniform scaling techniques which disregard formant and vowel identity and rely on single factors such as VT length, may not factor out all differences between children. We address this by comparing two formant scaling techniques, non-uniform (Lobanov) and uniform (DeltaF) (Johnson, 2020). 60 children, aged 4–10, produced /a, i, u/. Formants (F1–F3) were Lobanov- and DeltaF-scaled. Lobanov-scaled formants were more variable than Delta-F scaled formants, indicating that even after VT length normalization (uniform scaling), between-speaker variability remained in the children's speech. The children's F1:F2 ratio, a proxy for oral to pharyngeal cavity length ratio, explained some of this variability. This result suggests that children are not uniformly compensating for their morphologies and uniform scaling may not be appropriate at every developmental stage.

## Session 1aSCc

## Speech Communication: Speech Production II (Poster Session)

Authors will be at their posters from 11:15 a.m. to 12:00 noon

### *Contributed Papers*

**1aSCc1. Language effects on acoustic voice variation within and between talkers.** Yoonjeong Lee (Linguist, Univ. of California, 31-19 Rehabilitation Ctr., 1000 Veteran Ave., Los Angeles, CA 90095, yoonjeonglee@ucla.edu) and Jody Kreiman (Head and Neck Surgery, Univ. of California, Los Angeles, CA)

Acoustic voice spaces for English speakers are characterized mainly by variability in F0, the balance between higher harmonic amplitudes and inharmonic energy, and higher formant frequencies [*JASA* **146**(4), 3011 (2019)]. We extended this investigation to another language to test the hypothesis that a few biologically relevant measures will emerge commonly across languages, while remaining variance will depend on the structure of the language. This hypothesis was tested against sentence productions from 5 female and 5 male speakers of Seoul Korean. Like English, Korean does not have tone or phonation contrasts, but Seoul Korean exhibits specific phrase intonation patterns. PCAs were performed on scaled values of F0, formant frequencies, spectral noise, source spectral shape, and their variability, measured from vowels and approximants. Results revealed striking similarities between the acoustic voice spaces derived from Korean speakers and those for English speakers. For Korean voices, F0 and variability in lower formant frequencies (i.e., vowel quality) accounted for the most acoustic variance within and across talkers, presumably due to Seoul speakers' systematic use of these measures for phrasal/accentual information. These measures were insignificant for English voices. Our findings suggest that acoustic voice spaces are shaped by both biologically and phonologically relevant factors. [Work supported by NIH/NSF.]

**1aSCc2. Adding it all together: Temporally distributed minor correlates of a minor place contrast.** Stefon M. Flego (Linguist, Indiana Univ., Ballantine Hall 862, 1020 E. Kirkwood Ave., Bloomington, IN 47405, sflego@indiana.edu) and Kelly H. Berkson (Linguist, Indiana Univ., Bloomington, Bloomington, IN)

Laiholh (aka Hakha Chin, Hakha Lai) is a Kuki-Chin language that contrasts what has been described as lamino-dental and apico-alveolar articulations within coronal stops. As this type of minor place contrast is typologically uncommon and relatively understudied, the articulatory and acoustic properties distinguishing such pairs are not particularly well understood. It is unclear whether correlates reported for other languages hold cross-linguistically, or are language-specific. Our pilot investigation of this contrast in Laiholh revealed that very few acoustic differences between dentals and alveolars were stable across different vowel contexts and laryngeal settings. The current work expands our investigation by examining a larger data set and suite of acoustic parameters, including preceding and following formant transitions, closure duration, length of VOT, and spectral characteristics of aspiration. While acoustic differences tied to major place of articulation (labial versus coronal versus dorsal) are robust, those associated with the minor place difference within coronals are indeed minor. They have greater temporal distribution, however, being housed primarily in long-perseverating spectral information following stop release. We find constrained post-release spectral variation for apico-alveolar stops when compared with lamino-dental stops, suggesting that tongue body stabilization is associated with apical articulations, as reported for Wubuy (Best *et al.*, 2014).

**1aSCc3. Changes in the acoustic characteristics of speech in the later years of life.** Benjamin V. Tucker (Linguist, Univ. of AB, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, bvtucker@ualberta.ca), Mark L. Berardi, Eric Hunter (Michigan State Univ., East Lansing, MI), and Stephanie Hedges (Linguist, Univ. of AB, Edmonton, AB, Canada)

Acoustic and perceptual research has shown that an individual's vocal characteristics change over time [e.g., Harnsberger *et al.*, *J. Voice* **22**(1), 2334–2350 (2008)]. Most previous studies have investigated speech changes over time using cross-sectional data. The present study is a later-life longitudinal investigation of three speakers using publicly available archives of speeches given to large audiences on a semi-regular basis (generally with a couple of years between each instance). The group of speeches was given during the last 30–50 years of each speakers' life. From each speech, 5-minute samples (recordings and transcripts) were force-aligned to identify word and phoneme boundaries. Acoustic characteristics of the speech were extracted from the speech signal using Praat. In the present analysis, we investigate changes in the vowel space, pitch, word duration, segment duration, and speech rate. These acoustic characteristics are modeled using Generalized Additive Models [Hastie and Tibshirani, *Generalized Additive Models* (1990)] to allow for non-linear changes over time. The results are discussed in terms of vocal changes over the lifespan in the speakers' later-years. We find that not all of the effects are as expected and that the effects are idiosyncratic and dynamic over the lifespan.

**1aSCc4. Random forest classification of Gitksan stops.** Una Y. Chow (Linguist, The Univ. of BC, Totem Field Studios, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, una.chow@ubc.ca)

Gitksan, an endangered Tsimshianic language, has contrastive plain and ejective stops at five places of articulation (Rigsby, 1986). These lenis ejectives, which tend to be creaky-voiced, have been previously described as "implosives" (Hoard, 1978). This study asked which acoustic properties can distinguish ejective stops from voiced plain stops in Gitksan. Isolated Gitksan words ($n = 540$) containing initial prevocalic stops were audio-recorded from two male elder first-language talkers. A Praat script extracted voice onset time (VOT) and center of gravity (CoG) of the burst from the initial stop. VoiceSauce (Shue *et al.*, 2011) extracted voice quality measurements (difference in amplitude between five harmonic pairs, cepstral peak prominence, and harmonic-to-noise ratios across four frequency bands) and three formant transitions from the onset of the post-stop vowel. A random forest model (Breiman, 2001) was fitted with these 15 measures to determine their relative importance in predicting the stop types and their places of articulation. The results suggest that VOT is the primary and voice quality is the secondary cue to stop type, while CoG is the primary and formant transition is the secondary cue to place of articulation in Gitksan. Intertalker variation in voicing and glottalization affects the ranking of these cues.

**1aSCc5. Word frequency, predictability, and lexical class influence different aspects of Spanish tonic vowel production.** Scott J. Perry (Linguist, Univ. of AB, 3-28 Assiniboia Hall, Edmonton, ON T6G 2E7, Canada, sperry1@ualberta.ca), Matthew C. Kelley, and Benjamin V. Tucker (Linguist, Univ. of AB, Edmonton, AB, Canada)

The influence of lexical factors on speech production has not yet been investigated in Spanish to the same extent as for other languages such as English, French, Dutch, and German. Addressing this literature gap, the present study investigates how word frequency, predictability, and lexical class affect the duration and first two formant values of tonic monophthong vowels (hereafter vowels) produced by monolingual Spanish speakers from Madrid using the Nijmegen Corpus of Casual Spanish [Torreira and Ernestus, LREC'10 (2010), pp. 2981–2985]. A tonic vowel is defined as the most prominent vowel in a word's citation form. Word frequency and predictability based on the preceding word were calculated using the Spanish data from the OpenSubtitle corpus [Lison and Tiedemann, LREC'16 (2016), pp. 923–929]. Results of statistical modelling showed that the three aspects of vowel production under study were affected by all the examined lexical characteristics, although the effect sizes were small. Results are discussed in the context of previous work on Spanish vowels and cross-linguistic trends in speech production.

**1aSCc6. Speaking upside down: Manipulation of vowel formants with an inversion table.** Daniel Aalto (Commun. Sci. and Disord., Univ. of AB, 8205 114 St. NW, Edmonton, AB T6G2G4, Canada, aalto@ualberta.ca), Mike Fenner, Meagan Haarstad, Amberley Ostevik, Bill Hodgetts, and Jacqueline Cummine (Commun. Sci. and Disord., Univ. of AB, Edmonton, AB, Canada)

Posture affects the direction of gravitational forces on articulators. Since speakers seldom talk upside down, inverted posture offers an opportunity to observe the response of the neural control system to unfamiliar posture perturbations. Twenty native English speakers (16 women; 4 men) aged between 20 and 30 years gave written informed consent to participate in the study. They reported no history of hearing, balance, or speech problems. The participants were secured on an inversion table which was rotated and fixed to three positions: upright, supine, and upside down. The order of the postures was randomized. A headphone mounted microphone recorded series of ten repetitions of three words with target vowels /i,u,a/ embedded in a carrier word /hVd/ in each posture. Formants were extracted from the middle of the vowel using Praat. Results and discussion: Every participant's productions were impacted systematically by posture with greater acoustical effects observed for the unfamiliar posture. Based on mixed effects models, the F1 and F2 responses between participants aligned for /i/ in upside down condition compared to upright posture. The data suggests that unfamiliar (upside down) postural perturbation leads to greater deviation from its acoustic target than a familiar (supine) postural perturbation.

**1aSCc7. Acoustic differences between Salvadoran and Chilean Spanish [s].** Mariska Bolyanatz Brown (Spanish & French Studies, Occidental College, 1600 Campus Rd., Los Angeles, CA 90041, mbolyanatz@oxy.edu)

This paper examines the acoustic properties of [s] in Salvadoran and Chilean Spanish across multiple prosodic positions. Speakers are 36 working-class Chileans and Salvadorans balanced for dialect and gender who participated in an identical narration task in their respective home countries in 2015. Comparisons of fricative duration, relative intensity, the first four spectral moments, and percent voicelessness reveal multiple significant differences across dialects. Salvadoran Spanish [s] showed lower relative intensity, lower spectral Center of Gravity, higher variance, and more positive skewness across prosodic positions (word initial, word final, word-medial onset, and word-medial coda). Taken together, these results suggest previously unattested dialectal differences in degree of constriction and place of articulation of Spanish [s].

**1aSCc8. Aspiration in voiceless nasals in Angami.** Viyazonuo Terhiija (Phonet. and Phonology Lab, Central Library Bldg., Indian Inst. of Technol. Guwahati, Guwahati, Assam 781039, India, viyazonuo@iitg.ac.in) and Priyankoo Sarmah (Indian Inst. of Technol. Guwahati, Guwahati, Assam, India)

Angami or Tenyidie is a Tibeto-Burman language spoken in Nagaland, India, with contrasting voiced and voiceless nasals. Voiceless nasals are considered phonological sequences of voiced nasals followed by /h/ and phonetically as voiceless nasals (Bhaskararao and Ladefoged,1991). Using a Nasometer II 6450, we recorded 156 words with voiceless nasal onsets produced by Angami speakers. As the data was in stereo, it enabled us to separate the nasal and the oral channels. The results revealed that the initial 50–100 ms of a voiceless nasal is "nasal only," exhibited by high amplitude signals in the nasal channel. Afterward, aperiodic oral components appear resembling aspiration, while nasal components continue. Voicelessness in the 'nasal only' part depends on the context. For instance, /m̥$^{h}$a/, when embedded in the sentence frame as /a m̥$^{h}$a puba/, the nasal only part was fully voiced, and when produced in isolation, it was voiceless. The average nasalance of the aspiration following the nasal only part was higher when produced in isolation. Hence, we conclude, voiceless nasals in Angami are aspirated voiceless nasals, and voicing of the nasals in sentences is due to resyllabification. As [am.ha] with the nasal only part realized as voiced nasal and aspiration as a fricative.

**1aSCc9. A quantitative study of sound change in progress: The case of Suzhou Wu.** Wenxi Fei (The Education Univ. of Hong Kong, 10, Lo Ping Rd., Tai Po, New Territories, Hong Kong NA, Hong Kong, s1119503@s.eduhk.hk) and Albert Lee (The Education Univ. of Hong Kong, Hong Kong, Hong Kong)

Suzhou Wu is one of the major Chinese varieties and represents the Wu culture. Many researchers have anecdotally noted ongoing sound change of its phonemic and tonal systems. However, the latest investigation remains Ye and Sheng's description (1996). Thus, this study conducted an acoustic analysis on how young ($M = 21.63$, $SD = 0.92$) and middle-aged ($M = 46.75$, $SD = 3.77$) native speakers produced monosyllables in Suzhou Wu. Four male and four female local Mandarin-Wu bilinguals from each age group (16 speakers in total) were recruited to read 97 Chinese characters covering all phonemes and the 7 tones in the previous Suzhou Wu inventories (Wang, 1987; Ye and Sheng, 1996). Results suggest that there are noticeable acoustic differences between the two age groups in terms of consonants (excluding the lateral approximant [l]), vowels (some monophthongs: [i, a, ã, əʔ], most diphthongs and triphthongs), the tonal patterns (tone 2: low rising, tone 3: high falling). These results not only indicate sound change of Suzhou Wu, but also provide evidence for the age-related divergence which might be attributed to its frequent contact with the official language (Mandarin).

**1aSCc10. Regionally accented Mandarin lexical tones.** Yanping LI (MARCS Inst., Western Sydney Univ., Locked Bag 1797, Penrith, New South Wales 2751, Australia, yanping.li@westernsydney.edu.au), Catherine T. Best (tMARCS Inst., Western Sydney Univ., Sydney, New South Wales, Australia), Michael D. Tyler, and Denis Burnham (tMARCS Inst., Western Sydney Univ., Penrith, New South Wales, Australia)

This study investigated tone variations in regionally accented Mandarin (i.e., Standard Mandarin [SM] spoken by native speakers of other regional dialects in China). Yantai, Shanghai, and Guangzhou dialects were selected because their tone systems are different in various ways from the Beijing dialect, which is the basis of the SM tone system. 16 female regional speakers (4 speakers $\times$ 4 dialectal regions) were recruited to produce SM monosyllabic words that allow minimal contrasts among the four Mandarin lexical tones (i.e., level, rising, dipping, and falling tones). The overall *f0* contours within and across the four regional accents were modelled with growth curve analysis up to second-order orthogonal polynomials. The averaged tone shapes were significantly different within each of the regional accents, indicating that each group of regional Mandarin speakers successfully differentiated the four Mandarin lexical tones. However, the tone shape for each of the non-Beijing groups deviated significantly from Beijing Mandarin in two ways: (1) The quadratic term for the regional accents' dipping tones each differed significantly from Beijing accent; (2) The slopes of the

regional accents' rising and falling tones each differed significantly from Beijing accent. These two differences facilitate better understanding of tone variations triggered by regional accents.

**1aSCc11. Exploring invariant durational structure for voiceless and voiced stop quantity distinction in Kagoshima Japanese.** Shonosuke Koya (Graduate School of Humanities, Fukuoka Univ., 8-19-1 Nanakuma, Jonan-ku, Fukuoka 814-0180, Japan, britishgent.888@gmail.com), Shoji Ishibashi (Graduate School of Humanities, Fukuoka Univ., Fukuoka City, Fukuoka Prefecture, Japan), and Hajime Takeyasu (Faculty of Humanities, Fukuoka Univ., Fukuoka, Japan)

Hirata and Whiton [*J. Acoust. Soc. Am.* **118**, 1647–1660 (2005)] found that the durational ratio of stop closure to word (C/W ratio) is an invariant parameter for disyllabic words that differentiates voiceless single and geminate stops in Standard Japanese, a mora-timing language. The present study examines whether C/W ratio is an invariant parameter for voiced as well as voiceless stop quantity distinction in disyllabic words in Kagoshima Japanese, a syllable-timed dialect in Japanese. A production experiment was conducted in which three native speakers of Kagoshima Japanese (ranging from 54 to 86 in age) uttered in carrier phrases 27 disyllabic words, including 4 minimal pairs for voiceless stop quantity (e.g., popo versus poppo) and 4 minimal pairs for voiced stop quantity (e.g., kogo versus koggo). The rest of them were filler words. C/W ratio and five other possible parameters were calculated based on duration of each segment. Results of statistical analysis [classification accuracy, Akaike's information criterion (AIC), and Bayesian information criterion (BIC)] indicated that C/W ratio most accurately distinguishes single and geminate stops in Kagoshima Japanese as well as in Standard Japanese. [Work supported by Grant-in-Aid for Scientific Research, JSPS.]

MONDAY MORNING, 7 DECEMBER 2020 — 9:30 A.M. TO 11:00 A.M.

## Session 1aSPa

## Signal Processing in Acoustics and Underwater Acoustics: Random Matrix Theory in Acoustics I

Kathleen E. Wage, Cochair
*George Mason University, 4400 University Drive, Fairfax, VA 22151*

John R. Buck, Cochair
*ECE, University of Massachusetts, 285 Old Westport Road, North Dartmouth, MA 02747*

**Chair's Introduction—9:30**

### *Invited Papers*

**9:35**

**1aSPa1. Random matrix theory and underwater sound propagation.** Steven Tomsovic (Dept. of Phys. and Astronomy, Washington State Univ., Pullman, WA 99164-2814, tomsovic@wsu.edu) and Katherine Hegewisch (Geography, Univ. of Idaho, Moscow, ID)

Random matrix theory originated as the statistical theory of strongly interacting quantum systems, and has since been applied in an extremely broad range of applications. For example, it has been developed to accurately and efficiently model long range acoustic propagation in the ocean, including the construction of propagating acoustic time fronts. This problem can be viewed as dynamical symmetry breaking, i.e., the breaking of integrability, which has been explored in a variety of quantum mechanical contexts. The theory is expressed in terms of unitary propagation matrices that represent the scattering between acoustic modes due to sound speed fluctuations induced by the ocean's internal waves. The scattering exhibits a power-law decay as a function of the differences in mode numbers thereby generating a power-law, banded, random unitary matrix ensemble. In addition to its efficiency, the theory helps identify which information about the ocean environment can be deduced from the timefronts and how to connect features of the data to that environmental information. It also makes direct connections to methods used in other disordered wave guide contexts where the use of random matrix theory has a multi-decade history.

9:55

**1aSPa2. Modelling of sound propagation in the ocean using the matrix propagator.** Denis Makarov (Lab. of Nonlinear Dynamical Systems, POI FEBRAS, 43 Baltiyskaya St., Vladivostok 690041, Russian Federation, makarov@poi.dvo.ru)

The problem of sound propagation in an oceanic waveguide is considered. Attention is concentrated on properties of a wavefield propagator that governs transformation of an arbitrary wavefield in course of propagation. Using the basis of normal modes, we can represent the propagator in the matrix form. If horizontal inhomogeneity of a waveguide is weak, the propagator can be calculated by means of the first-order perturbation theory. In the case of random inhomogeneity, the apparatus of random matrix theory can be exploited. We demonstrate methods of propagator construction for realistic models of waveguides, including waveguides with adiabatic inhomogeneities and shallow-sea waveguides. A scheme of experimental measurement of a propagator in a shallow-sea waveguide is proposed.

## *Contributed Paper*

10:15

**1aSPa3. Modeling of signals scattered by turbulence on a sensor array using the matrix gamma distribution.** Vladimir Ostashev (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, vladimir.ostashev@colorado.edu), D. Keith Wilson, Matthew J. Kamrath (U.S. Army Engineer Res. and Development Ctr., Hanover, NH), and Chris L. Pettit (Aerosp. Eng., U.S. Naval Acad., Annapolis, MD)

The complex Wishart distribution provides a joint probability density for acoustic signals at a sensor array, in the regime of fully saturated scattering. In this presentation, the marginal distributions of the complex Wishart distribution are calculated using the characteristic function. A generalization of the complex Wishart distribution, namely the matrix gamma distribution, is proposed as a means for modeling the full saturation and unsaturated regimes. Asymptotic behavior of the matrix gamma distribution, and the consistency of its marginal distributions with predictions from the theory of wave propagation in random media, are studied. The predicted marginal distributions are compared to data from a comprehensive outdoor sound propagation experiment carried out in September 2018 at the National Wind Technology Center, Boulder, Colorado. In the experiment, microphones were attached to horizontal booms mounted at three heights on a 135-m meteorological tower, while a loudspeaker was positioned at the base of the tower or at a distance of 56 m from it.

## *Invited Paper*

10:35

**1aSPa4. Signal-to-interference plus noise ratio loss constrained robust adaptive beamformer inpsired by random matrix theory.** Christ D. Richmond (School of Elec., Comput., and Energy Eng., Arizona State Univ., P.O. Box 875706, GWC 316, Tempe, AZ 85287-5706, christ.richmond@asu.edu)

The optimal adaptive beamformer (ABF) maximizing output signal-to-interference plus noise ratio (SINR) has filter weights that depend on the data covariance and signal array response vector. The effectiveness of practical application of this optimal beamformer, however, is limited by (i) data stationarity (needed for covariance estimation), and (ii) knowledge of the true signal array response vector. Robust ABF attempts to address these two critical issues via a slight reformulation of the ABF problem, and often result in some form of diagonal loading yielding a hybrid beamformer that engages the tradespace between conventional beamforming (CBF) and ABF. The joint distribution of a CBF power spectral estimate and an ABF estimate based on the same data covariance reveals that estimates have a statistical coupling governed by the geometric cosine between their filter weights, i.e., the SINR loss between CBF and ABF. Thus, a robust ABF algorithm is proposed that constrains the SINR loss to an acceptable level while minimizing beamformer sensitivity to signal array response errors. This is practically appealing since this allows the user to specify the minimum SINR loss tolerable, and the resulting robust ABF solution uses the available degrees of freedom to reduce sensitivity to signal array response errors.

# Session 1aSPb

## Signal Processing in Acoustics and Underwater Acoustics: Random Matrix Theory in Acoustics II

Kathleen E. Wage, Cochair
*George Mason University, 4400 University Drive, Fairfax, VA 22151*

John R. Buck, Cochair
*ECE, University of Massachusetts, 285 Old Westport Road, North Dartmouth, MA 02747*

**Chair's Introduction—11:15**

### *Contributed Papers*

**11:20**

**1aSPb1. Whitening and source enumeration for large underwater arrays.** Jose A. Diaz-Santos (Elec. and Comput. Eng. Dept., George Mason Univ., 4400 University Dr., Fairfax, VA 22030, jdiazsan@masonlive.gmu.edu) and Kathleen E. Wage (George Mason Univ., Fairfax, VA)

Source enumeration for underwater arrays typically requires a whitening preprocessing stage due to ocean colored noise. A whitening transform requires the estimation of the noise-only sample covariance matrix (SCM). Estimation of a SCM is a challenging problem for large arrays in nonstationary environments due to the limited number of snapshots available for training. This talk considers a source enumeration method based on the dominant mode rejection (DMR) whitening transform (Diaz-Santos and Wage, 2015), inspired by Abraham and Owsley's DMR beamformer (1990). The DMR whitening transform reduces the number of snapshots required to estimate the noise-only SCM by replacing the small eigenvalues of the SCM with their average. In reduced-snapshot scenarios, replacing zero amplitude eigenvalues makes the SCM invertible. The new source enumeration method based on DMR whitening uses random matrix theory and high dimension, low sample size asymptotics to define a threshold that guarantees a specified probability of false alarm in scenarios where the number of array elements is larger than the number of noise-only snapshots. This talk analyzes the performance of whitened source enumeration method using noise from underwater experiments. [Work supported by ONR and NSWCDD.]

**11:40**

**1aSPb2. Random matrix theory analysis of the dominant mode rejection beamformer white noise gain.** Christopher Hulbert (George Mason Univ., 4400 University Dr., Fairfax, VA 22030, chulbert@gmu.edu) and Kathleen E. Wage (George Mason Univ., Fairfax, VA)

The dominant mode rejection (DMR) beamformer [Abraham and Owsley, *IEEE Oceans* (1990)] is an adaptive spatial processor that has been effectively used to null interference sources in scenarios where the number of training samples to estimate the sample covariance matrix (SCM) is limited. In these scenarios the ratio of the SCM dimension (array size) to the number of snapshots (training samples) is close to or even greater than one. Whereas classical asymptotic SCM results require the number of samples grow to infinity while keeping the dimension fixed, random matrix theory (RMT) provides a robust mathematical framework to analyze the SCM spectrum in snapshot-limited scenarios. RMT predictions of the SCM eigenvectors have already been applied to predict DMR beamformer notch depth in a single interferer environment (Buck and Wage, *IEEE SSP* (2012)]. This talk presents an RMT-based asymptotic approximation of the DMR SCM inverse in a loud multi-interferer environment. The RMT SCM model is used to predict the white noise gain (WNG) of the DMR beamformer, accounting for the impact when bulk spectral components are included in the dominant subspace. The performance of the RMT-based WNG prediction is compared to the sample mean using Monte Carlo simulations. [Funded by ONR 321US.]

**12:00**

**1aSPb3. Performance of the median dominant mode rejection beamformer against array element perturbations.** David C. Anchieta (ECE, Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, danchieta@umassd.edu) and John R. Buck (ECE, Univ. of Massachusetts Dartmouth, Dartmouth, MA)

The dominant mode rejection (DMR) beamformer [Abraham and Owsley, OCEANS (1990)] modifies the Capon's minimum variance distortionless response to operate with low rank sample covariance matrices (SCM). The DMR imposes an eigenstructure on the SCM by replacing the eigenvalues of the noise subspace with an estimated noise power. Standard DMR estimates the noise power by averaging the noise subspace eigenvalues. This estimated noise power is negatively biased when the dominant subspace dimension is overestimated. The proposed median DMR estimates the noise power from the median of the noise subspace eigenvalues, based on the Marchenko-Pastur probability distribution. The median estimator is not dependent on the estimated interferer subspace dimension. Simulations demonstrated that the median DMR improves the white noise gain (WNG) when compared to the standard DMR in snapshot deficient scenarios with overestimated interferer subspace dimension. Increased WNG also implies increased robustness to array perturbations [Gilbert and Morgan, BSTJ (1955)]. This work compared the median DMR to standard DMR in simulations with perturbed array element phase responses. The scenario included two interferers and background white noise. The median DMR preserved deeper notches than standard DMR in this scenario, increasing the output signal-to-noise ratio by roughly 1.1 dB. [Work supported by ONR code 321US.]

**12:20**

**1aSPb4. Compensating for adaptive beamformer overfitting with random matrix theory.** John R. Buck (ECE, Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, jbuck@umassd.edu)

Capon's Minimum Variance Distortionless Response beamformer requires the ensemble covariance matrix to compute the array weights [Capon, in Proc. IEEE (1969)]. In practice, adaptive beamformers compute their array weights from the estimated sample covariance matrix (SCM). In many scenarios, the environment changes too rapidly for the SCM to be full rank. The Dominant Mode Rejection (DMR) beamformer [Abraham and

Owsley, IEEE Oceans (1990)] mitigates this problem by imposing a structure on the eigenvalues of the SCM. DMR assumes that the background noise is spatially white, and enforces this assumption by replacing the weak SCM eigenvaues by their average. However, DMR still must estimate the appropriate subspace dimension for the dominant signals to minimize the expected beamformer output power. Estimating the DMR output power as a function of subspace dimension from the SCM results in a substantial negative bias for the output power caused by overfitting. Random matrix theory results on eigenvector fidelity allow us to approximate this bias in the output power, and compensate for the overfitting. This compensation yields a more accurate estimate of the beamformer output power as a function of subspace dimension. [Work funded by ONR 321US.]

MONDAY AFTERNOON, 7 DECEMBER 2020 1:05 P.M. TO 2:15 P.M.

# Session 1pAAa

## Architectural Acoustics, Noise and Speech Communication: Acoustics in Healthcare: Guidelines, Human Response, and the Way Forward Part III

Kenneth W. Good, Cochair
*Armstrong World Industries, Inc., 2500 Columbia Avenue, Lancaster, PA 17601*

Jay M. Bliefnick, Cochair
*Acentech, 33 Moulton St., Cambridge, MA 02138*

**Chair's Introduction—1:05**

### *Contributed Papers*

**1:10**

**1pAAa1. Potential restorative effect of the urban soundscape experience.** Hyun In Jo (Dept. of Architectural Eng., Hanyang Univ., 222 Wangsimni-ro, Seongdong-gu, Seoul 04763, South Korea, best2012@naver.com), Kounseok Lee (Dept. of Psychiatry, Hanyang Univ. Medical Ctr., Seoul, South Korea), and Jin Yong Jeon (Dept. of Architectural Eng., Hanyang Univ., Seoul, South Korea)

This study investigated the potential restorative effect of various urban soundscape experiences. The evaluation stimulus was implemented using virtual reality technology in a laboratory environment. Restoration-related semantic expression words were selected from the interview responses of 50 subjects, and the presence or absence of the expression of the words was newly proposed as a restorative potential criteria. The objective characteristics of urban soundscapes were analyzed based on auditory, visual, and object aspects, and subjects responded to sound source identification, perceived affective quality, and overall quality. The temperament and character inventory-revised short version and satisfaction with life scale were used to investigate individual characteristics. Based on this, a prediction model having an accuracy of 83% was proposed for the restorative potential criteria. As a result, it was found that the overall perception of soundscape was a crucial factor in increasing the restorative potential effect of the urban soundscape experience. In addition, it was confirmed that the restorative potential effect by individual characteristics varied significantly even in the same space. The findings of this study are expected to serve as a basis for sustainable and healthy urban soundscape design.

**1:30**

**1pAAa2. Virtual reality based assessment and education tool for auditory hallucination symptoms.** Junseok Won (Dept. of Medical and Digital Eng., Hanyang Univ., 222 Wangsimni-ro, Seongdong-gu, Seoul 04763, South Korea, redpig5808@naver.com), Kounseok Lee (Dept. of Psychiatry, Hanyang Univ. Medical Ctr., Seoul, South Korea), and Jin Yong Jeon (Dept. of Medical and Digital Eng., Hanyang Univ., Seoul, South Korea)

This study proposes a method to overcome the limitations of existing tests used to evaluate the symptoms of hallucinations in patients with schizophrenia. Moreover, it aimed to verify whether the disruption to daily life caused by hallucinations can be alleviated by performing a continuous attention task. To verify this, 44 patients with schizophrenia who experienced hallucinations and 22 general control groups were recruited. The patients were randomly assigned to the experimental group and patient control condition, and healthy people were assigned to the normal control condition. The experiment consisted of a total of six sessions: a pretest, four tests, and later tests. The experiment involved a 2-back task and auditory perception recognition, and only the experimental group and patient control group were provided with psychological education. The results of this study confirmed that hallucinations influenced the deterioration of attention-focused tasks. Furthermore, it was confirmed that the interference effect due to hearing loss was reduced by repeating the 2-back task and auditory recognition evaluation. Thus, the proposed hallucination evaluation and education method was effective in objectively evaluating hallucinations and reducing the disruption caused by hallucinations. In addition, the limitations of this study and suggestions for follow-up studies were presented.

1:50

**1pAAa3. Subjective perception of hospital environments with varying dynamic ranges of noise.** Jay Bliefnick (Acentech, 33 Moulton St., Cambridge, MA 02138, jbliefnick@gmail.com) and Erica E. Ryherd (Univ. of Nebraska, Omaha, NE)

Individuals may perceive noise stimuli differently based on the content of the source, due to factors such as the frequency spectra, level, temporal characteristics, or even the dynamic range of the noise source (the range of sound levels from quietest to loudest in a given sample). Results from earlier phases of this research indicated subject preference for quieter overall environments with a larger dynamic range of noise (based on the analysis of noise Occurrence Rates within real hospital units). In this phase of the study, a subjective perceptual laboratory test was designed and administered to 33 subjects to assess the annoyance perception of noise stimuli with varying dynamic ranges. Simulated hospital environments were utilized as the source stimuli, selected to represent a "typical" hospital unit patient area. It was found that in a controlled setting, subjects perceived noise stimuli with a wider dynamic range and louder peak noise events more negatively than noise sources with more consistent sound levels.



MONDAY AFTERNOON, 7 DECEMBER 2020 2:50 P.M. TO 4:20 P.M.

## Session 1pAAb

## Architectural Acoustics: Architectural Acoustics Potpourri I

Alaa Algargoosh, Chair
*Architecture, University of Michigan, 1123 McIntyre Dr., Ann Arbor, MI 48105*

**Chair's Introduction—2:50**

### *Contributed Papers*

2:55

**1pAAb1. Understanding the impact of room acoustic conditions on the kurtosis levels of various noise signals.** William J. Spallino (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 1110 S 67th St. #210C, Omaha, NE 68182, wspallino@huskers.unl.edu) and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE)

Impulsive noise can be common in certain occupational and recreational settings, such as manufacturing plants, construction sites, and firing ranges. While many regulations and guidelines for noise exposure exist, their mathematical basis is stronger for continuous noise, and concerns have been raised that impulsive noise may be more harmful to people's hearing than those metrics would let on. Much work has been done on establishing metrics that accurately assess the severity of risk associated with impulsive noise, and the kurtosis level shows promise. The effects of varying room acoustic conditions on that metric, however, have not been studied in great depth. This paper presents an investigation of how room acoustical properties may change measured kurtosis levels. Impulse responses of four different spaces were measured and used to assess each room's sound absorption and sound scattering properties. Four noise signals (a pure tone, white noise, Laplacian noise, and a series of impulses) were then computer-generated and convolved with each impulse response. The kurtosis level of each convolved signal was then calculated and analyzed with respect to both sound absorption and sound scattering properties. The potential relationships found between the kurtosis level and room acoustical properties are presented for each signal type.

3:15

**1pAAb2. Analysis of the acoustic characteristics of spaces relating to their emotional impact.** Alaa Algargoosh (Architecture, Univ. of Michigan, 2000 Bonisteel Blvd., Ann Arbor, MI 48109, alaas@umich.edu)

Sound and emotions have a link that has been widely studied in music. However, when sound propagates through space during a performance, the architectural features of the space impact how sound reflects, causing an alteration in some of the original sound characteristics. Therefore, it is essential to study how the acoustic environment contributes to the emotional impact. For instance, researchers in worship spaces have reported the significance of the acoustic environment in impacting emotions and enhancing the spiritual experience. Nevertheless, further research is required to understand the acoustic parameters that contribute to that effect. In a previous study, the researcher analyzed the emotional response to the acoustic environments of worship spaces using self-report and physiological indicators. This study builds on the earlier results by analyzing the acoustic characteristics of such spaces and their relationship to the emotional impact. Establishing such a link between room acoustics' parameters and emotions provides a guide for designing spaces that enhance the occupant's experience.

3:35

**1pAAb3. The effect of room acoustics on voice parameters: A systematic review.** Tomas E. Sierra (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign901 S. 6th St., Champaign, IL61820, tomases2@illinois.edu), Charles Nudelman (MGH Inst. of Health Professions, Boston, MA), Lady Catherine Cantor Cutiva (Dept. of Collective Health, Universidad Nacional de Colombia, Bogota, Colombia), and PASQUALE BOTTALICO (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

The purpose of this study was to quantify the relationships between voice parameters and room acoustics. A comprehensive literature search was conducted using PubMed/MEDLINE, Science Direct, and Scopus. Several terms regarding self-reported vocal parameters (vocal fatigue, vocal effort, vocal load, and vocal comfort), as well as possible variants, were included in the search. Also, we included different terms related to room acoustic parameters, such as reverberation time, noise conditions, decay times, room sizes, among others. Finally, a focus group of four experts in the field (current authors) worked together to make conceptual connections and quantify the proposed relationship. In total, 26 publications met the criteria for inclusion. The occurrence and frequency of the most common parameters in the literature are presented, and a quantitative summary of their relationships are reported. The most common journals and proceedings that may be involved in this subject are also pointed out. Through a comprehensive literature search, the most relevant parameters reported are Sound Pressure Level and self-reported vocal assessments for voice parameters, and reverberation time and/or noise conditions for room acoustics. These relationships are quantified and reported while maintaining the concepts as stated in the original articles and outlining their similarities.

3:55

**1pAAb4. Voice intensity changes in artificial acoustics.** Tomas E. Sierra (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign901 S. 6th St., Champaign, IL61820, tomases2@illinois.edu), Pasquale Bottalico, and Ella J Marzolf (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

The regulation of speech level is primarily affected by the physiological features of the speaker such as vocal tract size and lung capacity; however, the auditory feedback plays a fundamental role in voice production. The current study examined the effects of room acoustics in an artificial setting on voice production in terms of sound pressure level and the relationship with the perceived vocal comfort and control. Three independent room acoustic parameters were considered: reverberation time, gain (alteration of the sidetone), and background noise. An increase in the sidetone led to a decrease in vocal intensity, thus increasing vocal comfort and vocal control. This effect was consistent in different reverberation times considered. Mid-range reverberation times ($T30 \approx 1.3$ seconds) led to a decrease in the vocal intensity along with an increase on vocal comfort and vocal control. The presence of noise amplified the aforementioned effects for the variables analyzed.

MONDAY AFTERNOON, 7 DECEMBER 2020 — 1:05 P.M. TO 2:35 P.M.

## Session 1pAB

## Animal Bioacoustics: Innovative Tools for Animal Bioacoustics

Benjamin Hendricks
*SoundSpace Analytics, 2845 Penrith Ave., Cumberland, BC, Canada*

**Chair's Introduction—1:05**

### *Contributed Papers*

1:10

**1pAB1. A bio-inspired acoustic detector of mosquito sex and species.** Tim Ziemer (Bremen Spatial Cognition Ctr., Univ. of Bremen, Enrique-Schmidt-Str. 5, Bremen 28359, Germany, ziemer@uni-bremen.de), Julia Koch (Bremen Spatial Cognition Ctr., Univ. of Bremen, Bremen, Germany), Chaitawat Sa-Ngamuang, Myat Su Yin (Faculty of Information and Commun. Technol., Mahidol Univ., Salaya, Thailand), Mahmoud Siai (Bremen Spatial Cognition Ctr., Univ. of Bremen, Bremen, Germany), Benny Berkhausen (Faculty of Information and Commun. Technol., Mahidol Univ., Bremen, Germany), and Deniz Efe (Bremen Spatial Cognition Ctr., Univ. of Bremen, Bremen, Germany)

Mosquitoes mostly use their auditory system to recognize species and sex of other mosquitos. This has been proven based on behavioristic and neurologic experiments. This ability is amazing, since most of the frequencies that mosquitos create lie outside their own hearing range. It is evident that they do not respond to the frequencies themselves, but to nonlinear distortion products and combination tones between their own wing beat sound and the sound of other flying mosquitos. Based on mechanical, acoustical and neurological considerations, we are creating an analogue model of the mosquito's plumose antenna to calculate the afferent neural activity evoked by the scolopidia in the Johnston's organ when exposed to sound. The model successfully creates some of the nonlinearities that help mosquitoes identify sex and species of other mosquitoes. The model output can serve as a bio-inspired feature for automatic classification of mosquito sex and species. Applying machine learning with conventional audio features from the field of speech recognition and music information retrieval we observed good species discrimination abilities. However, we believe that bio-inspired features for mosquito species and sex recognition are more robust against environmental noise. Thus, they can contribute better to fighting malaria, dengue fever and other mosquito-borne diseases.

**1:30**

**1pAB2. *The Animal Audiogram Database*: A new resource for presenting and evaluating audiogram data on the web.** Denise Jäckel (Museum für Naturkunde, Invalidenstraße 43, Berlin 10115, Germany, denise.jaeckel@mfn.berlin), Alvaro Ortiz Troncoso, and Christian Bölling (Museum für Naturkunde, Berlin, Germany)

Knowledge of hearing abilities, as represented in audiograms, is vital for understanding animal acoustic physiology, behaviour, and ecology. Additionally, such knowledge plays an important role for measuring, predicting, or counteracting effects of anthropogenic noise on the environment. Currently, audiogram data is usually only available embedded in individual scientific publications and in various unstandardized formats, which makes access to and analysis of audiograms across sources cumbersome. We established a newly database that presents audiograms along with data on the audio-physiological experiments and original publications in a structured and easily accessible way. The interface enables combination of audiogram data for comparative analysis of different species, experimental conditions or publications. Focusing currently on marine vertebrates its content is the result of an extensive survey of the scientific literature and manual curation of the contained audio-physiological data. The database is designed to accommodate audiogram data on any biological group and purposed to be extended and serve as a reference source for audiogram data. It is publicly accessible at https://animalaudiograms.museumfuernaturkunde.berlin. [The database was developed as part of the project “Hearing in Penguins” funded by the German Environment Agency (UBA) with means from the Federal Ministry for the Environment, Nature Conservation and Nuclear Safety (BMU, FKZ3717182440).]

**1:50**

**1pAB3. Soundscape Viewer: An integrated toolbox of audio source separation for soundscape-based ecosystem assessment.** Yi-Jen Sun (Biodiversity Res. Ctr., Academia Sinica, 128 Academia Rd., Sec. 2, Nankang, Taipei 11529, Taiwan, elainesun442@gmail.com) and Tzu-Hao Lin (Biodiversity Res. Ctr., Academia Sinica, Taipei, Taiwan)

Soundscapes contain collective information on habitat quality and ecosystem dynamics. Current soundscape assessment methods, including clustering and ecoacoustic indices, may deliver unsatisfied performance when multiple sound sources are recorded simultaneously. We introduce an open-source Soundscape Viewer toolbox for separating soundscapes into sounds of biological, environmental, and anthropogenic sources. Based on non-negative matrix factorization (NMF), Soundscape Viewer enables audio source separation in both supervised and unsupervised manners. Conventional supervised NMF learns spectral features from labeled animal vocalizations, yet the performance is sensitive to noise levels. By assuming source-specific periodicity in unsupervised NMF, Soundscape Viewer can learn discriminative features between animal vocalizations and background noise. To cope with dynamic acoustic environments, Soundscape Viewer also integrates adaptive and semi-supervised learning approaches. Adaptive learning improves model generalization when sound characteristics differ from the training data, and semi-supervised learning allows the model to recognize new sound sources. We applied Soundscape Viewer in the evaluation of soundscape dynamics and the automatic detection of animal vocalizations. Our results reveal that the mutual interference between sound sources can be effectively reduced in various ecosystems. Source separation will facilitate biodiversity assessment and increase our understanding of biotic and abiotic sounds’ interactions.

**2:10**

**1pAB4. Nocturnal distribution of Guiana dolphins (*Sotalia guianensis*) in a coastal bay: Using the help of local fisherman and passive acoustic monitoring.** Mariana Barbosa (MAQUA, Rio de Janeiro State Univ., Rio de Janeiro, Brazil, marib66@gmail.com), Lis Bittencourt, Elitieri Santos-Neto, Tatiana Bisi, José Lailson-Brito, and Alexandre Azevedo (MAQUA, Rio de Janeiro State Univ., Rio de Janeiro, Brazil)

Overlap between dolphin distribution and fishing activities can lead to accidental by-catch. Assessing the nocturnal distribution of a resident population of dolphins can help highlight sensitive areas where by-catch occurs. The goal of this study was to use passive acoustics equipment deployed by local fisherman in different areas of Sepetiba Bay, southeastern Brazil, to assess Guiana dolphins’ distribution at night. In total, 64 recording sessions and 316 hours were collected at a sampling rate of 96 kHz, covering the period between 18:00 and 04:00. To detect Guiana dolphin sounds a Raven pro 1.5 band limited energy detector was defined between frequencies of 25 and 48 kHz and SNR>5dB. An experienced observer validated detections, quantifying according to time and area of occurrence. Guiana dolphins were more frequently encountered in two major areas, where they were present between 68% and 33% of recorded hours. The mean concentration of sounds/hour in the bay was 72 ($\pm$113), ranging from 0.6 to 325 across areas. In one location, there was also a higher concentration of detections between 18:00 and 19:00. Future work should rely on local knowledge of fisherman to detect major fishing areas during the night and help creating Guiana dolphin conservation plans.

# Session 1pAOa

## Acoustical Oceanography: General Topics in Acoustical Oceanography III

Nicholas A. Torres, Chair
*Mechanical Engineering, UT Austin, 301 E Dean Keeton St., Austin, TX 78712*

**Chair's Introduction—1:05**

### *Contributed Papers*

**1:10**

**1pAOa1. A yearlong record of ambient sound on the Chukchi Shelf.** Megan S. Ballard (Appl. Res. Labs.,Univ. of Texas, 10000 Burnet Rd., Austin, TX 78758, meganb@arlut.utexas.edu) and Jason D. Sagers (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Studies of ambient sound in the Arctic, both under ice and in the marginal ice zone, have spanned more than 50 years, but rapidly changing conditions with regards to declining ice cover have reduced the relevance of measurements taken in previous decades. Changes in the environment have resulted in changes in the ambient sound field, affecting both the sound generating mechanisms and the sound propagation. From October 2016 to October 2017, the Shallow Water Canada Basin Acoustic Propagation Experiment (SW CANAPE) was conducted on the Chukchi Shelf. One goal of CANAPE was to observe the changing soundscape. This talk presents acoustic recordings collected on the 150-m isobath with the Persistent Acoustic Observation System (PECOS), which contained a horizontal line array of hydrophones along the seabed and a vertical line array spanning a portion of the water column. This study examines the ambient sound level and uses k-means clustering to quantify the occurrence of six unique spectral shapes associated with different seasons and various sound generation mechanisms. The spectral clusters are correlated with environmental observations including sea concentration and thickness, wind speed, and air temperature. [Work supported by ONR.]

**1:30**

**1pAOa2. Geometrical modeling and analysis of low frequency acoustical scattering from cylindrically formed schools of swim bladder fish.** Luis Donoso (Inst. of Phys., Pontifical Catholic Univ. of Chile, Vicuna Mackenna 4860, Santiago, Region Metropolitana 7820436, Chile, lldonoso@uc.cl) and Christopher Feuillade (Inst. of Phys., Pontifical Catholic Univ. of Chile, Santiago, Metropolitan Region, Chile)

In previous work [*J. Acoust. Soc. Am.* **145**, 1654 (2019)], a dynamic computer model showed that, depending on interactions between individual fish, fish school behavior typically evolves into discoid, swarming, parallelized, or toroidal geometric forms. At low frequencies, the computed scattering behavior of these schools may be compared using a system of coupled differential equations, and also an "effective medium" method [*J. Acoust. Soc. Am.* **139**, 163–175 (2016)]. The latter technique treats a school as a single scattering object, where the internal medium has a complex propagation wavenumber dependent on the individual fish scattering properties. In the work here, the two-model comparison approach is extended to schools of cylindrical form, modeling them using a superspheroidal paradigm. A "T" matrix method is used to determine the scattering variations for the effective medium computations. The results show that, over much of the applicable frequency range, the scattering variations, as a function of frequency and angle, depend strongly on the specific geometrical shape and dimensions of the school, as well as the scattering properties of the fish constituting the school. This geometrical effect is particularly marked at very low frequencies, where the two models show close agreement. [Work supported by ONRG.]

**1:50**

**1pAOa3. Acoustic propagation in a seagrass meadow over diurnal and seasonal time scales.** Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758-4423, klee@arlut.utexas.edu), Megan S. Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Gabriel R. Venegas (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Austin, TX), Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Matthew C. Zeh (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX), Preston S. Wilson (Walker Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Abdullah F. Rahman (School of Earth, Environ., and Marine Sci., The Univ. of Texas Rio Grande Valley, Brownsville, TX)

Acoustic propagation in seagrass meadows is highly sensitive to oxygen production through photosynthesis. In addition to gas volumes encapsulated within the seagrass, free bubbles are released into the water column as oxygen diffuses through the plant tissue, affecting acoustic dispersion, absorption, and scattering. Because the oxygen production cycle is largely driven by sunlight, these effects exhibit a diurnal dependence. Previous work explored using acoustics to monitor seagrass photosynthetic activity, but this study presents new results that span both diurnal and seasonal time scales. Acoustic propagation experiments were conducted in a seagrass meadow in a shallow bay on the Texas Gulf Coast. A piezoelectric sound source transmitted frequency-modulated chirps (0.1–100 kHz) over several diurnal cycles, and the received acoustic signals were match-filtered to obtain band-limited impulse responses. Water temperature, salinity, depth, dissolved oxygen, and solar irradiance were concurrently measured with oceanographic probes. Measurements were taken both in winter and summer to examine the seasonal dependence of seagrass photosynthesis and its effect on the acoustic propagation environment. Dependence of the received acoustic signal on various environmental parameters will be discussed with the goal of using acoustics to study seagrass photosynthesis and productivity. [Work supported by ARL:UT IR&D and ONR.]

**2:10**

**1pAOa4. Finite element modeling of the low-frequency sound speed of seagrass leaves.** Nicholas A. Torres (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 301 E Dean Keeton St., Austin, TX 78712, natorres@utexas.edu), Megan S. Ballard, Kevin M. Lee, Gabriel R. Venegas (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Austin, TX), and Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Previous investigations related to acoustic propagation through seagrass have shown that sound waves are sensitive to the gas contained within seagrass tissue as well as free bubbles produced by photosynthesis. The

acoustical effects include reduced low-frequency sound speed and increased sound attenuation. The effects are more pronounced when the density of vegetation is higher as well as during daylight hours when sunlight-driven photosynthesis occurs. However, the application of mathematical models to describe these phenomena have been limited to effective medium models for water containing spherical gas bubbles. These approaches neglect both the effects of elastic properties of seagrass tissue as well as the shape of the aspherical gas bodies constrained within the plant. In this work, a finite element model of an acoustic resonator containing seawater and seagrass blades was developed to explain previously published measurements. The model utilizes independently measured values of the elastic properties of seagrass tissue and microscopic cross-section imagery of the gas volumes contained within the seagrass tissue. These results represent a step towards defining an appropriate effective medium model for acoustic propagation through seagrass tissue. [Work supported by ARL:UT IR&D and ONR.]



MONDAY AFTERNOON, 7 DECEMBER 2020 2:50 P.M. TO 4:20 P.M.

## Session 1pAOb

## Acoustical Oceanography: General Topics in Acoustical Oceanography IV

Jenna Hare, Chair
*Oceanography, Dalhousie University, 33-1173 Wellington St., Halifax, B3H3A2, Canada*

**Chair's Introduction—2:50**

### *Contributed Papers*

**2:55**

**1pAOb1. Vertical line array measurements of the sound radiated by melting glaciers in Hornsund Fjord.** Hari Vishnu (Acoust. Res. Lab., National Univ. of Singapore, 08-38, 54 Choa Chu Kang North 7, Singapore 689529, Singapore, harivishnu@gmail.com), Mandar Chitre (Acoust. Res. Lab., National Univ. of Singapore, Singapore, Singapore), Oskar Glowacki (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA), Mateusz Moskalik (Polish Acad. of Sci., Inst. of Geophys., Warsaw, Poland), M. Dale Stokes, and Grant B. Deane (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA)

Marine-terminating glaciers worldwide are melting rapidly in response to climate shifts, resulting in the delivery of freshwater into the oceans. Submarine melting at the glacier-water interface accounts for a significant component of the freshwater delivery from the glacier. This melting produces a distinct acoustic signature, providing a potentially viable modality to monitor glacial ice melting on a large scale via acoustic sensing. In order to evaluate the utility of ambient noise oceanography as a tool to quantify glacial ice-melt, in June 2019 we deployed a vertical hydrophone array and made acoustic measurements at some glacier termini in Hornsund Fjord, Spitsbergen. Quantification via array processing proves to be challenging due to (1) the space- and time-varying sound-speed profile in the underwater channel, and the way it refracts sound in an unknown manner, (2) limited vertical resolution of the array due to its aperture, and (3) interference from other noise sources such as melting icebergs, for example, contaminating the melt sound recordings. We present preliminary results from the processing which reveal different acoustic levels arising from submarine melting at different glaciers. The sound from the melt seems to be more dominant in the upper layers of the water at the glacier-sea interface.

**3:15**

**1pAOb2. Depth dependence of acoustic signals produced by bubble release events in melting glacier ice.** Hayden A. Johnson (Marine Physical Lab., Scripps Inst. of Oceanogr., 8622 Kennel Way, La Jolla, CA 92037, h3johnso@ucsd.edu), Hari Vishnu (Acoust. Res. Lab., National Univ. of Singapore, Singapore, Singapore), Grant B. Deane (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA), Phillip Tuckman (Massachusetts Inst. of Technol., Cambridge, MA), Oskar Glowacki (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA), and Mateusz Moskalik (Inst. of Geophys., Polish Acad. of Sci., Warsaw, Poland)

Melting glaciers and ice sheets are important contributors to sea level rise, but the rates at which they are losing mass are difficult to measure precisely. Using the acoustic signals produced by the rapid release of pressurized air from bubbles contained in the ice is a promising avenue for obtaining quantitative estimates of the rate of submarine melting at the termini of tidewater glaciers. The amplitude and character of the observed average signal generated by the bubble release events have been found to depend strongly upon the hydrostatic pressure of the water the bubbles are released into, and therefore on the depth at which said events occur; this dependence must be better understood before the goal of measuring melting rates can be achieved. Data from field experiments performed at the Hornsund fjord in Svalbard, Norway, in which the acoustic melting signal from blocks of glacier ice was recorded at varying depths, will be presented, and compared to the output of an idealized model for the acoustic signal produced by the release of bubbles from melting ice.

**3:35**

**1pAOb3. Remote acoustic measurement of the velocity within water-immersed gravity-driven granular flows.** Jenna Hare (Oceanogr., Dalhousie Univ., 33-1173 Wellington St., Halifax, NS B3H3A2, Canada, jenna.hare@dal.ca) and Alex E. Hay (Oceanogr., Dalhousie Univ., Halifax, NS, Canada)

Measuring bedload transport at high spatial and temporal resolution in energetic aqueous environments is challenging. Acoustic remote-sensing technologies are attractive because the measurement can be made without disturbing the mobile bed or the near-bed flow. Of particular interest is the development of broadband MHz-frequency acoustic systems capable of simultaneous measurements of backscatter amplitude and phase at mm-scale range resolution and 100 Hz sampling frequencies. Using such an instrument, we study granular flow in a water-submerged rectangular chute. By releasing sediments in the upstream portion of the chute, a O(1)cm-thick layer of avalanching sediment is produced. Trials were carried out for both erodible and fixed roughness beds. Natural sand and glass beads with median grain sizes ranging from 0.22 to 0.4 mm were used. The thickness of, and velocity profile within, the moving layer were measured using a wide bandwidth coherent Doppler profiler operating at 1.2 MHz. The velocity profiles are compared to estimates made with video imagery through the chute sidewall. The velocities at the sediment-water interface are compared to estimates made with a commercially available Doppler profiler (Vectrino) operating at 10 MHz and with imagery from a submerged video camera.

**3:55**

**1pAOb4. Statistical analysis and modeling of rain-generated ocean noise in the northeast Pacific Ocean.** Felix Schwock (Elec. and Comput. Eng., Univ. of Washington, 185 Stevens Way, Paul Allen Ctr. – Rm. AE100R, Seattle, WA 98195, fschwock@uw.edu) and Shima Abadi (Elec. and Comput. Eng., Univ. of Washington, Seattle, WA)

Large scale studies of underwater ambient noise during rainfall are important for assessing the ocean environment and enabling remote sensing of rainfall rates over the open ocean. In this study, we have evaluated approximately 3.5 years of acoustical and meteorological data recorded at the northeast Pacific Ocean continental shelf and slope. The acoustic data are recorded continuously at a sample rate of 64 kHz at 81 m depth and 581 m depth at the continental shelf and slope, respectively. The wind speeds and rainfall rates are provided by a surface buoy located in the vicinity of each hydrophone. Average power spectra have been computed for different rain rates and wind speeds, and linear and non-linear regression have been performed. The results are comparted between both measurement sites to evaluate the depth dependency of rain noise at the continental margin. In contrast to previous reports, we found that the rain noise levels between 100 Hz and 10 kHz are highly dependent on the prevailing wind speed. Our findings indicate that previously proposed algorithms for estimating rainfall rates from acoustic data are not universally applicable, but rather have to be adapted for different locations.

MONDAY AFTERNOON, 7 DECEMBER 2020 1:05 P.M. TO 2:35 P.M.

## Session 1pBAa

## Biomedical Acoustics and Signal Processing in Acoustics: Death to Delay and Sum: Advanced Beamforming III

Kevin J. Haworth, Cochair
*Internal Medicine, University of Cincinnati, 231 Albert Sabin Way, CVC 3939, Cincinnati, OH 45267-0586*

Kenneth B. Bader, Cochair
*Department of Radiology, University of Chicago, 5835 South Cottage Grove Ave., MC 2026, Q301B, Chicago, IL 60637*

**Chair's Introduction—1:05**

### *Contributed Papers*

**1:10**

**1pBAa1. Experimental results of tomographing the third-order acoustic nonlinear parameter.** Egor Kotelnikov (Acoust., Moscow State Univ., Leninskiye Gory, 1, V-1667, Moscow 119192, Russian Federation, eakotelnikov@ya.ru), Konstantin Dmitriev, and Olga Rumyantseva (Acoust., Moscow State Univ., Moscow, Russian Federation)

Tomography of the acoustic nonlinear parameters is a method of obtaining information about biological tissue using nonlinear interaction of acoustic waves. In the schemes under consideration, three plane coded non-collinear signals probe an investigated object and intersect inside it. Each point of the intersection region generates coded signal at the third-order combination frequencies. Correlation processing of received total signal allows determining the combination of the second- and third-order acoustic nonlinear parameters in each resolution element of the object. Results of physical experiment on reconstructing the spatial distribution of the combined acoustic nonlinear parameter by the scheme with one monochromatic and two coded probing signals are presented. Principal disadvantage of this scheme is nonlocality of generation area of the unique coded combination signal carrying information about the object. This makes it impossible to distinguish the signal generated by the only fixed resolution element of the object. As a consequence, reconstructed quantitative values of the nonlinear parameters can be distorted. The next step is the scheme with three coded probing signals, that is more complex for implementation. Nevertheless, now each unique signal is generated by the single resolution element. Thereby, it becomes possible to adequately reconstruct the desired values of nonlinear parameters.

**1:30**

**1pBAa2. Ultrasonic tomograph for reconstruction of spatial distributions of sound speed and absorption coefficient.** Dmitriy Zotov (Acoust., Moscow State Univ., Moscow, Russian Federation), Konstantin Dmitriev (Acoust., Moscow State Univ., Russian Federation, Moscow, Leninskie gory, 1, 2, Moscow 119992, Russian Federation, presentatio@mail.ru), and Olga Rumyantseva (Acoust., Moscow State Univ., Moscow, Russian Federation)

Ultrasound tomograph designed to reconstruct spatial distributions of sound speed and absorption coefficient in soft biological tissues (primarily in mammary gland) is presented. Distinctive feature of the tomograph is sparse annular antenna array with irregular arrangement of receiving-transmitting transducers. When an object is tomographed by its layers, 26 transducers in combination with antenna rotation allow providing the same data set as in case of nonrotating array with 256 transducers. Two-step processing algorithm uses both ray methods to reconstruct large-scale details of the required spatial distributions and wave methods to reconstruct small details. High degree of algorithm parallelism allows to reduce the time of experimental data processing by CUDA GPUs to several minutes per an image layer. Results of processing both model data and experimental data obtained for phantom objects are presented. Disadvantages of reconstruction at the first step of the algorithm with low resolution (0.5–1 cm) are analyzed. The high resolution (not worse than 1 mm), achieved at the second step of reconstructing the small details is illustrated. It is shown that these small details, which are the most informative for medical diagnostics at early stage of neoplasm, cannot be reconstructed without preliminary reconstruction of large-scale details.

**1:50**

**1pBAa3. Improving detection of bone surface defects using modified delay-multiply-and-sum reconstruction algorithm.** Philip M. Holmes (Mayo Clinic Graduate School of Biomedical Sci., 200 1st St. SW, RO_OS_02_2008, Rochester, MN 55902, holmes.philip@mayo.edu), Shawn O'Driscoll (Dept. of Orthopedic Surgery, Mayo Clinic, Rochester, MN), and Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN)

Osteochondritis Dissecans (OCD) is a joint defect in which a crack in the subchondral bone causes a separation of both subchondral bone and overlying cartilage from the bulk of the bone. Because there is an increased prevalence of OCD among youth athletes, an efficient, accurate, and non-ionizing screening method for this condition is being pursued. The aim of this project is to develop a clinical ultrasound image reconstruction algorithm that increases the detectability of bone surface defects, such as OCD. The proposed algorithm is a modified delay-multiply-and-sum (DMAS) algorithm that uses data acquired through synthetic aperture methods. The modifications made to the DMAS algorithm relevant to this work include transmit apodization and incoherent data summation. The modified DMAS algorithm was tested on data generated from k-Wave simulations and validated with experiments using bone mimicking material. Using both k-Wave and the bone mimicking material, we modeled fractures in the capitellum of the humerus with separations of 1, 2, and 3 mm. For experiments, we used synthetic aperture acquisitions using a Verasonics V1 system with a linear array (L7-4). The modified DMAS algorithm was found to enhance the clarity of bone surface defects when compared against DAS and DMAS algorithms.



**2:10**

**1pBAa4. Aberration correction in transcranial ultrasonic imaging using CT data and simulation-based focusing algorithms.** Sylvain Chatillon (LIST, CEA, Institut CEA LIST CEA Saclay, Bât. Digiteo - 565, Gif sur Yvette 91191, France, sylvain.chatillon@cea.fr), Arthur Waguet (LIST, CEA, Gif sur Yvette, France), Vincent Brulon (BIOMAPS, Orsay, France), Ekaterina Iakovleva (LIST, CEA, Gif sur Yvette, France), and Jean-Luc Gennisson (BIOMAPS, Orsay, France)

The complex structure of the skull bone, reflected in particular by spatial variations in thickness and density, leads to a heterogeneity of its acoustic properties. This results in a strong attenuation as well as specific phase shifts of the ultrasonic wave during its crossing, leading to a defocusing of the beam. The improvement of the quality of brain ultrasonography requires knowledge of these acoustic heterogeneities to shape the ultrasonic wavefront during the focusing and imaging processes. CT scan realized on a phantom of a human skull associated to a fusion tool available on an ultrafast ultrasound device (Aixplorer, Supersonic Imagine, France) allows the positioning of the phased array in real-time in the CT volume. Then, for a fixed position of the probe, the real geometry of the skull region insonified by the array can be used as input data of the simulation based imaging algorithms. In particular, the Plane Wave Imaging (PWI) and Total Focusing Method (TFM) algorithms exploit a direct model of beam propagation through 3-D complex surfaces in order to simulate and correct the aberrations due to the skull crossing. We illustrate the potential of this technique on several examples of brain echography.

# Session 1pBAb

## Biomedical Acoustics: General Biomedical Acoustics: Imaging I

Elisa Konofagou, Chair
*630 West 168th Street, Physicians & Surgeons 19-418, New York, NY 10032*

**Chair's Introduction—1:05**

### *Invited Paper*

**1:10**

**1pBAb1. Ultrasound super resolution imaging of nanodroplets with a multi-frequency hemispherical phased array.** Lulu Deng (Physical Sci. Platform, Sunnybrook Res. Inst., 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, lldeng@sri.utoronto.ca), Harriet Lea-Banks, Ryan Jones, Meaghan O'Reilly, Ran An, and Kullervo Hynynen (Physical Sci. Platform, Sunnybrook Res. Inst., Toronto, ON, Canada)

High resolution imaging of microvasculature is desirable for diagnostic and therapeutic applications in the brain. Here, we investigated the use of a 256-module sparse hemispherical transducer array to map the emissions of lipid-coated Decafluorobutane nanodroplets ($\sim$210 $\pm$ 80 nm, $10^7$–$10^9$ droplets/ml) flowing through tube phantoms (0.8 mm inner diameter). Each array module comprised 4 concentric cylindrical PZT-4 elements (55/306/612/1224 kHz). Droplets were vaporized at 55 kHz (0.10–0.18 MPa, 145 $\mu$s bursts every 2 s) and the resulting emissions were received on either the 306, 612 or 1224 kHz subarrays. Low-resolution 3-D images were formed using delay-and-sum passive beamforming, and super-resolved images were obtained via Gaussian fitting of the estimated point-spread-function to the low-resolution data. With super-resolution techniques, the mean lateral (axial) full-width-at-half-maximum image intensity was 35 $\pm$ 6 (67 $\pm$ 11), 16 $\pm$ 3 (32 $\pm$ 6), and 7 $\pm$ 1 (15 $\pm$ 2) $\mu$m from 160 (2970), 241 (2970), and 117 (4950) vaporization events (total frames), corresponding to $\sim$1/85 of normal resolution at 306, 612 and 1224 kHz, respectively. The mean positional uncertainties were $\sim$1/350 (lateral) and $\sim$1/180 (axial) of the receive wavelength in water. The pressure threshold for vaporization detection increased with increasing receive frequency. This study demonstrates the feasibility of mapping vaporized nanodroplets with passive beamforming and super-resolution imaging techniques.

### *Contributed Papers*

**1:30**

**1pBAb2. Effects of aberration on super-resolution ultrasound imaging using microbubbles.** Laura P. Peralta, Jemma Brown, Tiarna Lee (Biomedical Eng. and Imaging Sci., King's College London, London, United Kingdom), and Kirsten M. Christensen-Jeffries (Biomedical Eng. and Imaging Sci., King's College London, Imaging Sci., 4th Fl. Lambeth Wing, Westminster Bridge Rd., London SE1 7EH, United Kingdom, kirsten.christensen-jeffries@kcl.ac.uk)

Visualizating vasculature beyond the diffraction limit can be achieved using ultrasound super-resolution (USR). Typically, ultrasound (US) scanners model the target medium as homogeneous, assuming a constant speed-of-sound for time-of-flight based calculations. However, variations in US propagation velocity caused by varying tissue layers affect beamforming operations. This aberration is likely to have considerable effect on USR accuracy when imaging at depth. Here we investigate the effect of aberration on USR localization accuracy. Wave propagation through different tissue media was modelled from a linear transducer using k-Wave, and the resulting pulse-echo data from a point scatterer located at 45mm depth was beamformed. Media included a control homogeneous propagation medium, and aberrating media typical of liver imaging. Aberration effects were estimated using RMS arrival-time fluctuations (ATF) and energy-level fluctuations (ELF) at the scatterer location. Signals were extracted and localized with existing localization techniques. Results indicated USR localisation accuracy decreased with increasing aberration. Axial localisation errors reached 836.6 mm for an ATF 30.9 ns. Furthermore, errors increased with decreasing frequency from 4 to 3 MHz. The scale of these errors relative to micro-vascular structures of interest suggests that aberration will have considerable impact on USR performance and requires attention to ensure its success.

**1:50**

**1pBAb3. Coherence-based quantification of acoustic clutter sources in medical ultrasound.** James Long (Biomedical Eng., Duke Univ., 101 Sci. Dr., 1427 FCIEMAS, Durham, NC 27705, jc500@duke.edu), Will Long (Biomedical Eng., Duke Univ., Durham, NC), Nick Bottenus (Mech. Eng., Univ. of Colorado Boulder, Durham, NC), and Gregg Trahey (Biomedical Eng., Duke Univ., Durham, NC)

The relative contributions of aberration and incoherent noise sources, such as diffuse reverberation and thermal noise, to image quality in medical ultrasound are not well understood. We present a theoretical framework to predict losses in the correlation of ultrasonic backscatter, i.e., spatial coherence, and imaging contrast in response to combinations of incoherent noise and aberration levels. This framework is based on estimating the beamformer gain and channel SNR, using measurements of spatial coherence. A method to separate the contributions of incoherent noise and aberration in the spatial coherence domain is also explored and applied to predict losses in contrast. Results indicate excellent agreement between theory and simulations for beamformer gain and expected contrast loss. Coherence-predicted contrast loss and measured contrast loss in aberration differ by less than 1.5 dB on average, for a $-20$ dB contrast target and aberrators with RMS time delay errors and spatial autocorrelation profiles similar to those

measured *in vivo*. Results also indicate that the contribution of aberration to contrast loss varies with channel SNR, peaking around 0 dB SNR. This framework shows promise to facilitate novel, targeted clutter reduction strategies by providing a means to isolate and estimate the contributions of correlated and uncorrelated noise.

**2:10**

**1pBAb4. Occult regions of suppressed coherence in ultrasonic liver images.** Katelyn Offerdahl (Biomedical Eng., Duke Univ., 101 Sci. Dr., Durham, NC 27701, kro18@duke.edu), Matthew Huber, James Long, Will Long (Biomedical Eng., Duke Univ., Durham, NC), Nick Bottenus (Mech. Eng., Univ. of Colorado Boulder, Durham, NC), and Gregg Trahey (Biomedical Eng., Duke Univ., Durham, NC)

Lag-one coherence (LOC) estimates local levels of acoustic noise by measuring the spatial coherence between backscattered echo signals received by neighboring pairs of transducer elements. LOC can be directly related to signal to noise ratio (SNR) and contrast to noise ratio (CNR). We have acquired B-mode images and matched pixelwise LOC estimates in the livers of 10 healthy volunteers using a C5-2v probe on a Verasonics Vantage system and in Fullwave simulations with six abdominal walls over uniform speckle. We present evidence of temporally stable regions of suppressed LOC beneath the abdominal wall which recover to a stable asymptotic value at depth. Fullwave simulation results suggest that reverberation determines the initial amount of coherence suppression and aberration determines the asymptotic LOC value. The *in vivo* LOC values beneath the abdominal wall range from roughly 0.4 to 0.85, corresponding to SNRs of −3.5 dB to 15 dB, and the length of coherence suppression ranges from 0.5 cm to 2.5 cm. These regions are occult; they present as temporally stable uniform liver on B-mode images. This is significant because clinicians will not be aware that lesions may be much more difficult to detect in this region, potentially leading to missed diagnoses.



MONDAY AFTERNOON, 7 DECEMBER 2020 — 2:50 P.M. TO 4:20 P.M.

## Session 1pBAc

## Biomedical Acoustics: General Biomedical Acoustics: Imaging II

Elisa Konofagou, Chair
*630 West 168th Street, Physicians & Surgeons 19-418, New York, NY 10032*

**Chair's Introduction—2:50**

### *Contributed Papers*

**2:55**

**1pBAc1. Motion estimation for ultrasound image sequences using deep learning.** Skanda Bharadwaj (Comput. Sci. and Eng., Penn State Univ., W103, Westgate Bldg., University Park, State College, PA 16802, ssb248@psu.edu) and Mohamed Almekkawy (Comput. Sci. and Eng., Penn State Univ., State College, PA)

Increased number of medical images being used for medical praxis makes automatic processing of images a necessity. Conventional techniques of motion estimation in Ultrasound images such as exhaustive search-based block matching (ES-BM) are known to be computationally expensive and are unsuitable for portable devices. On the other hand, research in computer vision has helped the development of deep learning-based techniques for real-time motion estimation of day-to-day non-medical objects. In this paper, we propose to adopt one such deep neural network-based Fully Convolutional Siamese tracker along with Linear Kalman Filter (SiamFC-LKF) to track regions of interest in ultrasound image sequences. Siamese networks use two convolutional neural networks to create a feature map of the given reference block and the possible candidate blocks of subsequent frames. The candidate block of the subsequent frame with the maximum correlation value is considered as the tracked output. Since SiamFC does not consider any motion model, we introduce a linear Kalman filter to track the wall of the carotid artery. SiamFC-LKF and ES-BM were tested on five different image sequences of the longitudinal section of the carotid artery. Our experiments showed that SiamFC-LKF was 6 times faster and performed better than ES-BM in most of the cases.

**3:15**

**1pBAc2. Wave field diversification using multiple frequency bands for compressive ultrasound imaging.** Aiguo Han (Univ. of Illinois at Urbana-Champaign, 306 North Wright St., Urbana, IL 61801, han51@illinois.edu)

Ultrasound image reconstruction as an inverse program is often ill-conditioned due to the uniformity of the ultrasonic wave field. Recently, a hologram approach has been introduced to break the uniformity of the wave field, enabling compressive ultrasound imaging using a single-element transducer. In this approach, the surface of a single-element transducer is covered by an acoustic hologram, a thin layer of a coded plastic mask, to scramble the phase of the wave field. Mechanical translation or rotation of the hologram has been used to further increase wave field diversity. However, mechanical control of the hologram could potentially limit the practical application of the method. In this study, we propose to increase wave diversity by dynamically changing the frequency band, without hologram translation or rotation. Computer simulations were performed to evaluate the effectiveness of the proposed method. The 3-D wave fields were simulated for a single-element transducer covered by a randomly generated acoustic hologram. The frequency band of the transmitted pulse was varied. The spatial-temporal wave field diversity was evaluated in terms of spatial variance and pixel-pixel correlation. The simulation results showed that multiple frequency bands increased the wave field diversity desirable in compressive ultrasound imaging. (NIH Support: R01CA226528, R01DK106419, R01HD089935)

3:35

**1pBAc3. Real-time, patient-adaptive ultrasonic intensity adjustment: Hepatic imaging observations.** Matthew Huber (Biomedical Eng., Duke Univ., 101 Sci. Dr., 1427 FCIEMAS, Durham, NC 27705, matthew.huber@duke.edu), Katelyn Flint, James Long, Will Long (Biomedical Eng., Duke Univ., Durham, NC), Nick Bottenus (Mech. Eng., Univ. of Colorado Boulder, Durham, NC), and Gregg Trahey (Biomedical Eng., Duke Univ., Durham, NC)

Increasing B-mode ultrasound transmit intensity improves signal to noise ratio (SNR) by increasing backscattered echo magnitude relative to thermal noise. To ensure ultrasound remains safe, acoustic output limits exist, and regulatory bodies advise observing the ALARA (As Low As Reasonably Achievable) principle. Despite this, studies show sonographers rarely adjust intensity, resulting in unnecessary acoustic exposure or sub-optimal image quality. We have developed a framework for automated transmit intensity adjustment which we demonstrate on a Verasonics Vantage ultrasound system and C5-2v transducer. The coherence of signals received by neighboring ultrasound array elements, the lag-one coherence (LOC), quantifies clutter and temporally varying noise and serves as the automation feedback parameter. In the automated sequence, receive data are quickly acquired over a region of interest (ROI) for nine intensities ranging from mechanical indices (MI) of 0.08 to 1.4. LOC asymptotically increases with acoustic intensity as the effect of thermal noise decreases until intensity increases minimally improve SNR; the intensity at 98% of the maximum LOC is used for B-mode scanning. In preliminary hepatic imaging studies, a ROI of 7 lateral lines extending 30ö£ axially achieves temporally stable intensity updates. The optimization time for this ROI is 0.7 seconds, enabling real-time intensity adaptation.

3:55

**1pBAc4. Acousto-optic imaging using focused and plane wave ultrasound pulses.** Lukasz J. Nowak (Univ. of Twente, Drienerlolaan 5, Enschede 7522 NB, The Netherlands, l.j.nowak@utwente.nl) and Wiendelt Steenbergen (Univ. of Twente, Enschede, The Netherlands)

Acousto-optic imaging exploits the effects of modulation of light by acoustic waves propagating in an optically scattering medium to determine its optical properties. The imaging resolution is determined by the acoustic pressure field distribution inside the sample. In the ideal case, this distribution should be limited to a relatively small, strictly defined volume within which the light modulation occurs. Practical applications of this technique impose specific requirements on transducers configuration which constitute additional limitations on shaping the acoustic pressure field characteristics. We investigate the possibilities of improving performance of an acousto-optic imaging system utilizing a linear ultrasound array by comparing results obtained using plane wave and focused pulses and different apodization patterns. Acoustic pressure field distributions were determined numerically and with hydrophone measurements. We introduce an ultrasound focus quality coefficient in order to describe and compare the confinement of the expected light modulation volume. Acousto-optic images were obtained experimentally by measuring contrast decrease of the interference patterns of a laser light transmitted through samples and recorded by a camera. The results show that for focused pulses the off-focal pressure field components have significant influence on the determined values, and that this issue can be mitigated by using plane wave imaging.

MONDAY AFTERNOON, 7 DECEMBER 2020 — 2:50 P.M. TO 4:20 P.M.

## Session 1pEA

## Engineering Acoustics: General Topics in Engineering Acoustics II

Thomas E. Blanford, Cochair
*The Pennsylvania State University, State College, PA 16804*

Caleb F. Sieck, Cochair
*Code 7160, U.S. Naval Research Laboratory, 4555 Overlook Ave. SW, Washington, D.C. 20375*

Michael R. Haberman, Cochair
*Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758*

**Chair's Introduction—2:50**

### *Contributed Papers*

2:55

**1pEA1. Reduction of structural vibration and radiated noise using acoustic resistive volume.** Jin Liu (Carrier Global Corp., Carrier PKWY, Syracuse, NY 13221, jin.liu2@carrier.com)

Undesirable noise levels and shell vibrations from pipes and cavities with internal pulsating pressures is a common problem for HVAC, automobiles and airplanes. This study is concerned with providing effective noise control solutions without affecting overall system performance. Acoustic impedance optimization is carried out using a Fluid-Structure-Interaction FEA solver to reduce radiated noise by adjusting the acoustic impedance in the pipes and cavities to avoid excitation of internal acoustic modes. Porous materials with high porosity (>0.9) and with hole sizes much smaller than the viscous penetration depth are introduced inside the pipe and cavities to create resistance volumes to increase acoustic dissipation. In this way, acoustic traveling waves are promoted and the formation of standing waves is minimized. This in turn reduces surface vibration levels of the pipes thereby reducing radiated noise.

**3:15**

**1pEA2. An experimental capability to capture azimuthal variation of high-frequency noise in automotive centrifugal Compressors.** Pranav Sriganesh (The Ohio State Univ., 930 Kinnear Rd., Columbus, OH 43212, sriganesh.1@osu.edu), Rick Dehner, and Ahmet Selamet (The Ohio State Univ., Columbus, OH)

Stringent emission and fuel economy standards have led to the widespread use of turbocharged spark-ignition internal combustion engines in light-duty cars and trucks. However, advancements in impeller design and casing treatments to improve compressor performance and efficiency have introduced high-frequency noise. In particular, the elevation in tonal noise at the blade pass frequency (BPF) has become a significant contributor to the total sound pressure level in some of the modern automotive compressors. Due to high compressor rotational speeds, BPF noise occurs at relatively high frequencies where sound may propagate as multi-dimensional acoustic waves within the compressor ducting. The present work describes a compact experimental setup to examine the azimuthal non-uniformity of noise with the help of a rotating compressor inlet duct on a steady flow turbocharger gas stand. The rotating duct houses dynamic pressure transducers to capture time-resolved, in-duct acoustic pressure at different azimuthal locations. This experimental setup thereby facilitates the comparison of acoustic pressures at these angular locations for identical compressor steady flow operating conditions, while also scanning the compressor map of a contemporary turbocharger.

**3:35**

**1pEA3. Multidomain modeling of moving coil speakers including multiple nonlinearities.** Stephen C. Thompson (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, sct12@psu.edu)

Accurate modeling of moving coil speakers must include nonlinearities in the magnetic force factor, mechanical mounting stiffness, and magnetic circuits. The same model must also include complicated linear effects such as eddy currents in the magnetic structure and thermoviscous losses in enclosures and small transmission paths. Including all of these in a single model is beyond the capability of conventional analog circuit models. Multidomain modeling with Simscape or Modelica offers a method of solution for all the features mentioned. A Simscape model with several nonlinearities will be demonstrated and compared with measurements from prototype speakers.



**3:55**

**1pEA4. Novel approach in loudspeaker test stimulus generation using digital methodologies.** Riccardo Balistreri (Eng., Garmin, 5 Wilkins St., Auckland 1011, New Zealand, riccardo.balistreri@fusionentertainment.com)

Noise is the most widely used type of signal for acoustic power testing of loudspeakers and audio equipment. It is normally characterised by crest factor, "coloration" and filtering, or power spectral density, for the purpose of approximating the spectral content of music to suite a Device Under Test (DUT). The accustomed way to have equipment generate these signals such as IEC 268-5, EIA 426-B, or pink noise with specified crest factor and bandwidth for use on a DUT, is to first generate white noise, filter by 3dB/oct to get pink noise, then adjust the crest factor by clipping to specified value (some standards show a diagram using diodes), and further filtering, to match the required bandwidth or spectral distribution. This practice is in the opinion of the authors, antiquated. Music is born by natural vibrations and follows more of a gaussian distribution with time varying characteristics and the above-mentioned equipment and processes alter its noise distribution and have little or no time varying characteristics. This paper will place a study on music spectral content, distribution and time varying characteristics and explore synthesis of new signals for loudspeaker and audio equipment testing focusing on easy generation within the digital domain using computers.

## Session 1pNSa

# Noise and Education in Acoustics: Larry H. Royster Memorial Session I

Elliott H. Berger, Cochair
*Berger Acoustical Consulting, 221 Olde Mill Cove, Indianapolis, IN 46260*

Noral D. Stewart, Cochair
*Stewart Acoustical Consultants, 7330 Chapel Hill Rd., Suite 201, Raleigh, NC 27607*

**Chair's Introduction—1:05**

### *Invited Papers*

**1:10**

**1pNSa1. Larry Royster—The early years, North Carolina regional chapter, and scholarship support.** Noral D. Stewart (Stewart Acoust. Consultants, 7330 Chapel Hill Rd., Ste. 201, Raleigh, NC 27607, norals2020@sacnc.com)

This paper in three parts will discuss (1) the early life of Larry Royster, his education and early career to the point that he became involved in hearing conservation, (2) the North Carolina Regional Chapter of the Acoustical Society of America which he founded, and (3) his strong support of scholarships for students especially the Royster Award administered by the regional and student chapters of the society.

**1:30**

**1pNSa2. Larry Royster's influence on my life and on noise and hearing conservation, as told by one of his students.** Elliott H. Berger (Berger Acoust. Consulting, 221 Olde Mill Cove, Indianapolis, IN 46260, eberger@compuserve.com)

I first met Larry Royster as a master's degree student in search of a thesis project and research funding, having heard that he was looking for a student to work on a potential grant concerning occupational hearing loss. The day I entered his office at NC State was one of the most impactful of my life, dramatically influencing the trajectory of my university work and professional career. Working with Larry gave me the opportunity to manage and conduct a research project in the textile industry, gain insights into the hearings at the Department of Labor that were to shape the Hearing Conservation Amendment, and to see first-hand how Larry influenced and guided the North Carolina DOL approach to noise and hearing conservation. He was there when I had my first postã graduate school job offer from E-A-R Division of Cabot Corporation, later Aearo Technologies and 3M, and then became an important friend and collaborator for much of my subsequent research. He chaired ANSI and NHCA working groups and received the ASA Silver Medal and other important society awards. Larry was a friend, mentor, and dedicated scientist whose impact on noise and hearing conservation will be the topic of my talk.

**1:50**

**1pNSa3. Forty-six years of learning, experiencing, and enjoying work with Larry Royster.** Dennis Driscoll (Driscoll Acoust., 2560 S. Orchard St., Lakewood, CO 80228, dennis@driscollacoustics.com)

It was late August 1973, on my first day as an undergraduate student, when I meet my academic advisor—Dr. Larry H. Royster. As I crammed 4 years of education into the next 5 years Larry was always there to provide counsel and guidance. Then I started grad school and it turned out he also had a research assistantship available to study occupational hearing loss. Two years later I completed my graduate program under Larry's direction. My professional career started in 1980 at Standard Oil Corporation of Indiana. Eight years later I decided to become a consultant in occupational noise, which I remain to this day. Throughout my career Larry continued to mentor me. We worked together on many different ventures from teaching professional development courses, working on machinery noise control, hearing loss litigation support, to co-authoring publications. My talk will describe many of these professional experiences. However, it is the personal relationship and fun I shared with Larry that I cherish the most, and I am pleased to share with you, too. To say Larry shaped my professional life is an understatement – overuse of hyperbole not withstanding meeting Larry was the turning point in my life.

2:10

**1pNSa4. Roysters and high fidelity hearing protection.** Mead Killion (none, 61 Martin Ln., Elk Grove Vlg, IN 60007, eberger35@comcast.net)

The Roysters' Chicago-Symphony-Orchestra study led to abandoning Plexiglass Barriers between the brass and viola sections: They reduced exposure by 3 dB in the viola section but increased it 3 dB for the bassoons. That study was also an important factor in the successful introduction by Etymotic Research of (a) 15 dB custom Musicians Earplugs (designed by ASA fellow Elmer Carlson), (b) inexpensive ready-fit ER20 20 dB HiFi earplugs (designed with Berger and Falco and Stewart), and (c) MusicPRO earplugs. About the latter, John Yeh, clarinetist with the CSO said: "These electronic earplugs allow me to hear everything, including the Maestro's instructions, while damping the onslaught of the brass band." Nonetheless, there were frequent academic concerns about "not enough protection." Indeed, E-A-R marketing initially declined to offer the 20 dB earplugs, and the Common Market outlawed low-attenuation earplugs and dictated that orchestras should not play above 85 dBA SPL. (I once measured a 106 dB SPL SLM peak at a balcony seat in Orchestra Hall!) These "not enough" complaints were wonderfully answered by Royster (1993): "Of the 1/3 of all factory workers who need protection, 76% need less than 10 dB." He also observed that too much attenuation led to non-use.



MONDAY AFTERNOON, 7 DECEMBER 2020 2:50 P.M. TO 4:20 P.M.

## Session 1pNSb

## Noise and Education in Acoustics: Larry H. Royster Memorial Session II

Elliott H. Berger, Cochair
*Berger Acoustical Consulting, 221 Olde Mill Cove, Indianapolis, IN 46260*

Noral D. Stewart, Cochair
*Stewart Acoustical Consultants, 7330 Chapel Hill Rd., Suite 201, Raleigh, NC 27607*

**Chair's Introduction—2:50**

### *Invited Papers*

2:55

**1pNSb1. Population-based age-adjustment tables for use in hearing conservation programs.** Gregory Flamme (SASRAC, 2264 Heather Way, Forest Grove, OR 97116, gflamme@sasrac.com) and Kristy K. Deiters (SASRAC, Forest Grove, OR)

Starting in the 1970s, Larry Royster and his colleagues conducted a series of large-scale studies demonstrating that people with non-Hispanic Black race/ethnicity had better hearing sensitivity than their non-Hispanic White counterparts. One major outcome of this work was the adoption of specific age-related hearing sensitivity expectations by race in the original version of ANSI S3.44. Royster's accomplishments informed our work over the last decade, the goal of which was to develop age-adjustment tables for replacing those used in current U.S. regulations. The old age adjustment tables, which are frequently mislabeled age "corrections", were developed by NIOSH based on cross-sectional observations available in the 1970s and included relatively few people. We developed nationally representative cross-sectional age trends based on 9,937 audiograms and validated them for men using 76,185 exam records from 9,340 noise-exposed workers. Results indicated that the 1970s-era age adjustments overestimate current trends, which is consistent with multiple studies showing an overall decline in the prevalence of hearing loss within the United States. The results also indicate a substantial reduction in age-related changes among people with non-Hispanic Black race/ethnicity. Our work confirmed and extended Royster's observations in a large regional sample to the U.S. population as a whole.

**3:15**

**1pNSb2. Larry H. Royster memorial session.** Richard L. Neitzel (Environ. Health Sci., Univ. of Michigan, 1415 Washington Heights, 6611 SPH I, Ann Arbor, MI 48109, rneitzel@umich.edu)

Larry Royster conducted pioneering research on the effectiveness of occupational hearing conservation programs, with a particular focus on the performance of hearing protection devices (HPDs) and audiometric database evaluation. Dr. Royster also conducted important early research into non-occupational noise exposures such as sporting events, recreational activities, and air travel. This presentation will briefly review key findings from Dr. Royster's work on these topics from the 1970s–1990s, and will then summarize the findings of a number of recent related studies. These will include: an evaluation of hearing conservation program effectiveness and hearing protector performance among aluminum manufacturing workers; an assessment of the contributions of occupational and non-occupational noise to hearing loss risk in a large sample of individuals in New York City; and a review of patterns and trends in occupational noise exposures in the US from 1979–2013. Collectively, these studies indicate that the risk of hearing loss from noise remains substantial among US workers, as well as in the general public that participates in noisy leisure-time activities, and highlight the importance of continuing the research that Dr. Royster started in order to protect the hearing health of Americans.

**3:35**

**1pNSb3. Remembering remarkable Larry H. Royster.** Julia D. Royster (none, 4706 Connell Dr., Raleigh, NC 27612, jsd.royster@gmail.com)

Larry Royster was a dedicated scientist and engineer whose recognized contributions in the area of noise control and hearing conservation were by no means his only interests. After retiring, Larry's amazing mind was free to follow his fancy into a wide variety of topics. He measured countless phenomena and graphed the results, read nonfiction prodigiously and rated what he read, and wrote summaries for the public about what he was learning. He took up golf and amateur radio with zeal. Liberated from responsibilities that he had always taken quite seriously, he teased his friends and laughed with them. He also reflected on qualities of character that drove him throughout his teaching and research career. In this presentation I will share sides of Larry that were not obvious to many.

MONDAY AFTERNOON, 7 DECEMBER 2020 — 1:05 P.M. TO 2:35 P.M.

## Session 1pPAa

## Physical Acoustics: General Topics: Infrasound

Brice Coffer, Chair

*North Carolina State University, North Carolina State University, Campus Box 8208, Raleigh, NC 27695*

**Chair's Introduction—1:05**

### *Contributed Papers*

**1:10**

**1pPAa1. Infrasound measurements of tornadoes and other severe storm events at close range.** Brandon White (Mech. and Aerosp. Eng., Oklahoma State Univ., 201 General Academic Bldg., Oklahoma State University, Stillwater, OK 74078, brandon.c.white@okstate.edu), Bryce Lindsey, Imraan Faruque, and Brian R. Elbing (Mech. and Aerosp. Eng., Oklahoma State Univ., Stillwater, OK)

Recent experimental evidence suggests that, during tornadogenesis and through the life a tornado, acoustic waves at frequencies below human hearing (infrasound) are produced. To date, acoustic data required to identify the fluid mechanism responsible for this infrasound has been limited—often gathered by large fixed installations. The design and deployment of a mobile Ground-based Local Infrasound Data Acquisition (GLINDA) system was completed at Oklahoma State University to expand the number of samples and enable close-range measurements which would mitigate the measurement uncertainty associated with long distance atmospheric propagation. GLINDA has been deployed alongside Oklahoma-based media storm chasers since May 2020 and has already returned data over multiple severe weather events, including tornadic measurements acquired with GLINDA on 22 May 2020 in Lakin, Kansas. The GLINDA-equipped storm chaser vehicle can additionally provide acoustic data on other weather events such as wildfires and winter weather storms as the vehicle is appropriate for sustained, close observation of these environments. This presentation will cover system design, measurement capabilities, and acoustic production by select weather phenomenon.

### 1:30

**1pPAa2. Infrasound characteristics of nontornadic and tornadic thunderstorms using high-resolution simulations.** Brice Coffer (North Carolina State Univ., Campus Box 8208, Raleigh, NC 27695, becoffer@ncsu.edu) and Matthew Parker (North Carolina State Univ., Raleigh, NC)

There has been increased interest in improving severe weather detection by supplementing NOAA's WSR-88D radar network with an infrasound observation network, which may be able to detect distinct subaudible signatures from tornadic thunderstorms. While there is evidence that tornadic thunderstorms exhibit observable infrasound signals, what is not well-understood is whether these infrasound signals are unique to tornadic storms compared to nontornadic storms or whether there is any useful signal prior to tornadogenesis (which would be most relevant to NWS forecasters). In this presentation, we will present the results of high-resolution simulations of severe thunderstorms, specifically tailored to represent acoustic wave propagation with frequencies of 0.1 to 1 Hz. The spectral analysis of pressure perturbations generated by a nontornadic and tornadic supercell thunderstorm will be compared against each other. Two additional sensitivity tests that employ different microphysics schemes (i.e., the representation of precipitation within the storm) will be discussed, since prior work has indicated that the melting of precipitation is a dominant contributor to infrasound generation in thunderstorms. So far, preliminary results suggest that there are no discernible differences in infrasound observed in the vicinity of tornadic versus nontornadic supercell thunderstorms prior to tornadogenesis.

### 1:50

**1pPAa3. Prediction of long-range infrasound propagation from tornadoes based on new atmospheric boundary layer wind tunnel experiments.** Tianshu Zhang (Mech. and Aerosp. Eng., Univ. of Florida, 939 Sweetwater Dr., Gainesville, FL 32611, zhang.tianshu@ufl.edu), Steven A. Miller (Mech. and Aerosp. Eng., Univ. of Florida, Gainesville, FL), Mariel T. Ojeda, and Kurtis Gurley (Civil & Coastal Eng., Univ. of Florida, Gainesville, FL)

Tornadoes generate infrasonic noise that propagates over long distances. NOAA is developing tornado early warning systems via infrasound microphone arrays. Accurate detection must be coupled with a propagation algorithm that accounts for infrasound distortion due to the effects of atmospheric turbulence. Numerical solutions of the generalized Burgers' equation and the acoustic ray equations are used to predict changes to infrasonic tornado noise within the turbulent atmosphere. The effects of nonlinearity, refraction, attenuation, dispersion, and scattering are captured by the solver. We define a new parameter to model the alteration of infrasound that depends on atmospheric turbulence. Turbulent kinetic energy, integral length scale, integral time scale, and local speed of sound are arguments of the model and vary along each ray path. A series of tests are designed and conducted at the University of Florida Boundary Layer Wind Tunnel to calibrate and validate the model at scale. Predictions show satisfactory agreement with measurements with varying wind speed, turbulence intensity, humidity, and temperature. Finally, we present a parametric study for propagation of sinusoidal signals at 90 dB and 7 Hz in regions of the United States using realistic weather.

### 2:10

**1pPAa4. Severe weather observations from a deployable four sensor infrasound array.** Christopher Petrin (Mech. & Aerosp. Eng., Oklahoma State Univ., 139 Adv. Technol. Res. Ctr., Stillwater, OK 74078, christopher.e.petrin@gmail.com), Real J. KC, and Brian R. Elbing (Mech. & Aerosp. Eng., Oklahoma State Univ., Stillwater, OK)

Infrasound is sound at frequencies below 20 Hz. It has been observed to be emitted by tornado-producing storms up to two hours before tornadogenesis. Due to the low atmospheric attenuation of sound at these low frequencies, infrasound may be detected several hundreds of kilometers away under ideal atmospheric conditions. Therefore, passive infrasound monitoring has potential for the study and prediction of tornadoes and other severe weather if the received infrasound signals can be correlated with thermodynamic and flow field properties of the storms and/or tornadoes. Previous work accomplished at Oklahoma State University has focused on observations from a single stationary array located in Stillwater, OK. However, the large distance between this array and the nearest NEXRAD II radar stations makes low-level storm characterization challenging to accomplish in tandem with infrasound observation of severe weather. Therefore, a second array was designed to be deployable at various sites in Oklahoma within 50 km of the radar stations, in order to ensure low-altitude radar resolution. Containing four microphones, the new array was deployed during the summer of 2020 at several sites affiliated with the Oklahoma Mesonet. Details of the array's design and deployment will be presented, as well as preliminary data collected during non-tornadic thunderstorms in 2020. [This work was supported by NOAA Grant NA19OAR4590340 and NOAA Grant NA18OAR4590307.]

# Session 1pPAb

## Physical Acoustics: General Topics: Potpourri I

Aaron Gunderson, Chair
*Applied Research Laboratories, The University of Texas at Austin, 10,000 Burnet Rd., Austin, TX 78758*

**Chair's Introduction—1:05**

### *Contributed Papers*

**1:10**

**1pPAb1. Modeling and estimation with multiple reflections.** Samiya Alkhairy (MIT, 77 Mass Ave., 54-317, Cambridge, MA 02139, samiya@alum.mit.edu)

When medium property variation is not smooth and slow or perturbative, multiple reflections are non-negligible. We model multiples and develop associated methods for estimating medium parameters from received signals. We focus on 1-D which is fundamental theoretically and has direct applications to layered media, e.g., nondestructive testing for delamination in layered composites. We account for all reflection multiples: intra and inter-layer including sub and supra-source layer multiples. We represent the system in two ways, and develop associated model expressions and estimation methods. For the block diagram representation, we employ system functionals to develop closed-from expressions of the transfer function that do not require integration. Using this expression, we can estimate parameters by optimization. For the tree-diagram representation, we develop expressions that construct the response semi-incrementally in time. We develop non-optimization-based methods to estimate parameters from these expressions without prior knowledge of the number of layers. For broad-band signals, this method runs in real-time. Our models and methods account for various multiples, are easily modifiable for various configurations (of pre-source layers and location of source-receiver) and source radiation patterns (uni/bi-directionality). Configurational flexibility, diagrammatic representations, simple closed-form expression, interpretable models, and form of estimation methods are strengths of our models and methods.

**1:30**

**1pPAb2. Frequency-dependent analytic models for scattering off finite half planes.** Samiya Alkhairy (MIT, 77 Mass Ave., 54-317, Cambridge, MA 02139, samiya@alum.mit.edu)

Established theories of wave scattering off points and infinite reflectors have limitations for many problems encountered in wave propagation. Our goal is to extend acoustic scattering theory towards a model that not only describes conventional scattering but also the less idealized case of scattering off finite objects. We develop an analytic object-centric forward frequency domain model of the signal at the receiver. The expression is explicitly parameterized in terms of medium and finite object properties that explicitly incorporates frequency-dependence. Our construction uses perturbation theory and integrates over effective point sources along the object surfaces that acts as finite half planes. From this, we derive closed-form expressions using integral approximation methods; and we also determine frequency, material and geometry conditions under which the conventional theories and associated operators apply. We utilize limiting cases to guide and test our model. The model also provides nontraditional extensions of reflection theory, e.g., for sources with spherical radiation patterns. The unified treatment of the continuum of wave phenomena is not only important theoretically but may also be useful in frequency-based object-centric estimation, array optimization and designing signal processing and acquisition schemes.

**1:50**

**1pPAb3. Nonlinear reflection coefficient of finite amplitude waves.** Aakash Khandelwal (Mech. Eng., Michigan State Univ., East Lansing, MI), Prathamesh Bilgunde, Daniel Barnard (Ctr. for Nondestruct. Eval., Iowa State Univ., Ames, IA), and Sunil Kishore Chakrapani (Elec. and Comput. Eng., Michigan State Univ., 428 S. Shaw Ln., Rm. 2120 Eng. Bldg., East Lansing, MI 48824-1226, csk@egr.msu.edu)

Ultrasonic waves incident at the interface between two linear media will be reflected based on the linear reflection coefficient, which depends on the linear acoustic impedances of the media. However, propagation of ultrasonic waves in quadratically nonlinear media can lead to nonlinear reflection coefficients which are dependent on the frequency, source amplitude, and nonlinearity of the two mediums. The objective of this work is to explore the existence of a nonlinear reflection coefficient between two mediums using analytical, numerical, and experimental methods. For simplicity and consistency, a water-solid interface was chosen and finite amplitude ultrasonic waves between 2.25 MHz and 25 MHz were used to study the interfacial response. The analytical results show the existence of a nonlinear RC, but whose effect is very minimal. Experimental and numerical results show that the transfer of energy in harmonic generation affects the RC more than other factors. The influence of several factors like interaction zone, shock generation, etc. will be discussed.

**2:10**

**1pPAb4. Nonlinear harmonic imaging of solids.** Guillermo E. Huanes-Alvan (Mech. Eng., Michigan State Univ., East Lansing, MI) and Sunil Kishore Chakrapani (Elec. and Comput. Eng., Michigan State Univ., 428 S. Shaw Ln., Rm. 2120 Eng. Bldg., East Lansing, MI 48824-1226, csk@egr.msu.edu)

Nonlinear ultrasonics has been shown to be sensitive to small scale damage and microstructural changes in solids. However, measuring the nonlinearity parameter of a solid requires complex calibrations and point-by-point measurements using contact transducers. This poses a challenge when it comes to applying this technique for imaging. The present work explores the use of immersion based nonlinear ultrasonics to perform nonlinear harmonic imaging of solids. A solid sample immersed in liquid (water) will form a "3-layer" structures (water-solid-water) when two ultrasonics transducers are used in through transmission mode. This work will explore the analytical modeling of such a structure using the KZK equation to model the combined effect of attenuation and diffraction of the three layers. The objective of the modelling effort is to develop an inversion model to invert the acoustic nonlinearity parameter from experimental results. The nonlinear model was further validated using experimental measurements carried out on a Steel sample with localized damaged.

## Session 1pPAc

## Physical Acoustics: General Topics: Bubbles I

Alicia Casacchia, Chair
*The University of Texas at Austin, Austin, TX 78758*



**Chair's Introduction—2:50**

### *Contributed Papers*

**2:55**

**1pPAc1. Time-domain simulation of acoustic wave scattering and internal propagation from gas bubbles of various shapes.** Jiacheng Hou (Mech. and Aerosp. Eng., Utah State Univ., 4130 Old Main Hill, Logan, UT 84322-4130, jiachenghou@aggiemail.usu.edu), Zhongquan C. Zheng (Mech. and Aerosp. Eng., Utah State Univ., Logan, UT), and John S. Allen (Mech. Eng., Univ. of Hawai'i at Mānoa, Honolulu, HI)

Acoustic scattering and resonances of incident waves from a gas bubble are simulated using a time-domain simulation based on numerical solutions of the conservation laws. The time histories of scattering pressure and velocity, both outside and inside the bubble, are obtained simultaneously from an immersed-boundary method facilitating the investigation of both exterior and interior fields for complex geometries. The acoustic resonances of the bubble are investigated for various bubble sizes, shapes and inner gas parameters and compared to the partial wave scattering solutions for spherical bubbles. Agreement is shown with the analytical solutions in the linear acoustic limit. The resonance frequency increases with the bubble's inner background pressure and independent of the incident wave amplitude or frequency. In addition to scattering outside of the bubble, acoustic propagation inside the bubble is investigated with respect to the monopole resonance. A significant advantage is that this time-domain simulation combined with the immersed-boundary method can be readily adapted for various shapes of bubbles including oblate or prolate. The scattering and resonance behaviors are simulated and compared with the analytical results involving a shape factor and extended for previously less investigated shapes of significance to underwater and physical acoustics applications including "pancake shaped" bubbles.

**3:15**

**1pPAc2. Numerical study on evolution of weakly nonlinear waves into an acoustic soliton in bubbly water.** Takahiro Ayukai (Eng. Mech. and Energy, Univ. of Tsukuba, Tsukuba, Japan) and Tetsuya Kanagawa (Eng. Mech. and Energy, Univ. of Tsukuba, 1-1-1 Tennodai, Tsukuba, Ibaraki 305-8573, Japan, kanagawa.tetsuya.fu@u.tsukuba.ac.jp)

Finite but small amplitude (i.e., weakly nonlinear) propagation of pressure (or acoustic) waves in an initially quiescent water uniformly containing many spherical microbubbles is numerically studied. Especially, an effective equation for the case of low frequency long wave is the Korteweg–de Vries–Burgers (KdVB) equation, which consists of the nonlinear, dissipation, and dispersion terms. Two major solutions of KdVB equation are a shock wave (i.e., dissipative case) and a soliton (i.e., dispersive case). The purpose of this study is the elucidation of spatio-temporal evolution of these three terms (not coefficients) and that of resultant waveforms. The results are summarized as follows: (a) a solitary pulse waves were formed by the nonlinear and dispersion terms and well-known nonlinear interaction was observed; (b) the number of formed solitons was dependent on the initial waveform such as Gaussian, shock, and rectangular waveforms; (c) the number of formed solitons was strongly dependent on the initial void fraction.

**3:35**

**1pPAc3. Weakly nonlinear theory on a thermal effect inside bubble on plane pressure waves in bubbly water.** Takafumi Kamei (Eng. Mech. and Energy, Univ. of Tsukuba, Tsukuba, Japan), Tetsuya Kanagawa (Eng. Mech. and Energy, Univ. of Tsukuba, 1-1-1 Tennodai, Tsukuba, Ibaraki 305-8573, Japan, kanagawa.tetsuya.fu@u.tsukuba.ac.jp), and Takahiro Ayukai (Eng. Mech. and Energy, Univ. of Tsukuba, Tsukuba, Japan)

Especially focusing on a thermal conduction and a thermodynamic process inside the bubble, we theoretically analyze weakly nonlinear propagation of plane progressive pressure waves in an initially quiescent water uniformly containing many spherical microbubbles. We utilize the famous model proposed by Prosperetti (1991) as the energy conservation equation. Some temperature gradient model such as Preston *et al.* (2002) and Sugiyama *et al.* (2005) are also used. For simplicity, the viscosity of gas phase and the phase change and mass transport across the bubble–liquid interface are omitted. We then derived the Korteweg–de Vries–Burgers (KdVB) equation including a correction term without differentiation due to the consideration of thermal conduction, from the basic equations for bubbly flows based on a mixture model. As a result, we clarified that the thermal effect contributes not only the dissipation effect but also the nonlinear effect. The coefficient of nonlinear term includes the ratio of specific heats of the ideal gas inside the bubble. We then concluded that the thermodynamic process inside bubble determines the wave propagation process.

**3:55**

**1pPAc4. Theoretical analysis on a nonlinear interaction of plane pressure waves with long and short wavelengths in bubbly water.** Katsunori Tanaka (Eng. Mech. and Energy, Univ. of Tsukuba, Tsukuba, Japan) and Tetsuya Kanagawa (Eng. Mech. and Energy, Univ. of Tsukuba, 1-1-1 Tennodai, Tsukuba, Ibaraki 305-8573, Japan, kanagawa.tetsuya.fu@u.tsukuba.ac.jp)

Weakly nonlinear interaction between short and long waves in an quiescent water uniformly containing many spherical gas bubbles is theoretically investigated under the assumption of one-dimensional progressive pressure wave propagation. For simpicity, the bubbles do not coalesce, break up, disappear, and appear; thermal conduction and phase change are neglected. By the use of the singular perturbation analysis based on the method of multiple scales, we conduct the perturbation calculations up to the fifth order. We then derive the coupled equations the so-called SKdV type equations, from the set of governing equations for bubbly flows, one is the nonlinear Schrödinger (NLS) type equation for the short wave with a nonlinear interaction term and the other is the Korteweg–de Vries (KdV) type equation for the long wave with a nonlinear term of the short wave. Finally, some comments for waveforms will be presented from both viewpoints of analytical and numerical.

## Session 1pSAa

## Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics: Acoustic Metamaterials I

Christina J. Naify, Cochair
*Naval Research Lab, 4555 Overlook Ave. SW, Washington, D.C. 20375*

Alexey Titovich, Cochair

Bogdan-Ioan Popa, Cochair
*Univ. of Michigan, Ann Arbor, MI 48109*

**Chair's Introduction—1:05**

### *Invited Paper*

**1:10**

**1pSAa1. Phonon manipulation with valley-Hall junctions.** Jihong Ma (Univ. of Vermont, Burlington, VT), Kai Sun (Univ. of Michigan, Ann Arbor, MI), and Stefano Gonella (Univ. of Minnesota, 500 Pillsbury Dr. SE, Minneapolis, MN 55455-0116, sgonella@umn.edu)

In this work, we investigate experimentally the existence of in-plane valley Hall edge states in hexagonal lattices with relaxed space inversion symmetry and we exploit them to realize non-trivial backscattering-free interfaces. We propose special tessellations of lattice domains in which multiple non-trivial domain walls coalesce to form junctions, and we study how guided waves behave when they impinge on such junctions. We show that, through a proper selection of the interface types and orientations, it is possible to achieve junctions with different and exotic wave manipulation capabilities. Specifically, we discuss two applications. The first is a direction-selective splitter, which endows its host lattice with asymmetric wave transport characteristics. The second is a signal delayer that is realized by embedding in the lattice a non-trivial waveguide loop, along which the energy can be temporarily trapped and periodically released. These junctions can serve as building blocks for a new structural logic paradigm enabled by topological mechanics.

### *Contributed Papers*

**1:30**

**1pSAa2. Enhanced acoustic transmission through an aqueous acoustic metasurface.** Amber M. Groopman (Code 7160, NRC Res. Associateship Program, U.S Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, amber.groopman.ctr@nrl.navy.mil), Theodore P. Martin (Excet, Inc., Springfield, VA), Charles A. Rohde (Code 7160, U.S. Naval Res. Lab., DC, WA), Caleb F. Sieck, and Matthew D. Guild (Code 7160, U.S. Naval Res. Lab., Washington, DC)

Acoustic metasurfaces are ultrathin (subwavelength thickness) structures that can demonstrate extreme acoustic properties, such as negative or near-zero values, similar to those seen in acoustic metamaterials. The negligibly small thickness of acoustic metasurfaces make them a more practical alternative to metamaterials for certain applications. In this work, we experimentally demonstrate an aqueous acoustic metamaterial that employs subwavelength, flexural elements that acoustically act in parallel to achieve enhanced acoustic transmission and broadband negative effective density. The metasurface was constructed from a brass plate, which was machined to have a parallel arrangement of circular, flexural elements on the surface, and experimentally tested in water over the nominal range of 50–100 kHz. The reflected and transmitted acoustic data were compared between a brass plate with and without the circular flexural elements. It was observed that the aqueous acoustic metasurface made from a brass plate with circular flexural elements showed improved broadband transmission. A detailed description of the experimental testing and analysis will be discussed. [Work supported by the Office of Naval Research]

**1:50**

**1pSAa3. Nonlinear dispersive waves in 2-D micropolar lattices.** Samuel P. Wallen (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, sam.wallen@utexas.edu) and Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Architected materials, also known as mechanical metamaterials, are solid structures designed by leveraging geometry, rather than composition, to achieve desired mechanical properties. Past studies on mechanical metamaterials have demonstrated properties that are rare or absent in naturally occurring materials, such as negative Poisson's ratio, chirality, and extremely high ratios of bulk to shear modulus and strength to weight. Most of the existing literature on architected materials has been focused on linear behavior, though some works have incorporated nonlinearity, often leveraging mechanical instabilities to re-configure a lattice. In the present work, we explore nonlinear dispersive waves in 2-D architected lattices with rotating microstructure, which are man-made analogues of nonlinear micropolar continua. Specifically, we consider a discrete model that consists of a periodic array of rigid bodies connected by nonlinear springs, focusing on the weakly nonlinear dynamic regime. For wavelengths much longer than the characteristic lengths of the microstructure, we demonstrate that the

nonlinear behavior of longitudinal and transverse acoustic wave modes may be described by evolution equations of the Kadomtsev-Petviashvili type, which include effects of weak nonlinearity, weak dispersion, and weak transverse variation. Solutions of these evolution equations will be discussed and compared with direct numerical simulations of the discrete lattice.

**2:10**

**1pSAa4. Coupled-mode analysis of a nonreciprocal elastic wave circulator.** Benjamin M. Goldsberry (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, bgoldsberry@utexas.edu), Samuel P. Wallen, and Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Acoustic and elastic metamaterials with time- and space-dependent material properties have received great attention recently as a means to increase the degree of control over mechanical waves. A circulator is a device that can transmit and receive signals in a nonreciprocal fashion using a network of ports attached to a junction having either spatiotemporally varying properties or momentum bias. The work presented here considers the numerical study of an elastic wave circulator, which is composed of three elastic waveguides attached to a thin elastic ring, creating a three-port network with 3-fold rotational symmetry. Nonreciprocity is achieved for both flexural and extensional waves by modulating the elastic modulus of the ring in a rotating fashion. An approximate model based on coupled-mode theory, which makes use of a plane wave basis, is derived and implemented. Steady state solutions are found, which include the generated harmonics of the modulation frequency. The coupled-mode model is compared with a finite element approach, and conditions on the system parameters that enable a high degree of nonreciprocity are then discussed in terms of a set of characteristic dimensionless numbers. [Work supported by NSF EFRI.]



MONDAY AFTERNOON, 7 DECEMBER 2020 — 2:50 P.M. TO 4:20 P.M.

## Session 1pSAb

## Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics: Acoustic Metamaterials II

Christina J. Naify, Cochair
*Naval Research Lab, 4555 Overlook Ave. SW, Washington, D.C. 20375*

Alexey Titovich, Cochair
*Naval Undersea Warfare Ctr., Carderock Division, West Bethesda, MD 20817-5700*

Bogdan-Ioan Popa, Cochair
*Univ. of Michigan, Ann Arbor, MI 48109*

**Chair's Introduction—2:50**

### *Contributed Papers*

**2:55**

**1pSAb1. Effective metamaterial properties for non-local active acoustic media.** Nathan Geib (Univ. of Michigan, 1587 Beal Ave. Apt. 13, Ann Arbor, MI 48105, geib@umich.edu), Aritra Sasmal (Univ. of Michigan, Los Angeles, CA), Bogdan-Ioan Popa, and Karl Grosh (Univ. of Michigan, Ann Arbor, MI)

Acoustic metamaterials are generally composed of complex arrays of subwavelength unit cells that can be cumbersome to analyze and simulate. Consequently, it is common practice to approximate metamaterials as homogeneous media with unique material properties, most commonly negative or near-zero density, negative or near-zero bulk modulus, or both negative or near-zero density and bulk modulus. In addition to easing the computational burden associated with complex metamaterial structures, effective material properties can provide deeper understanding of system behavior, providing additional insight and tools in the development of acoustic metamaterials. Here, we present a method for determining effective material properties for a non-local, active acoustic metamaterial. We contrast these properties with those typically associated with acoustic metamaterials and discuss how such properties are characteristic of a new class of acoustic metamaterials.

**3:15**

**1pSAb2. Metamaterial lens for real-time ultrasound beam steering using single-channel scalar control.** Hyung-Suk Kwon (Dept. of Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109, kwonhs@umich.edu), Bogdan I. Epureanu, and Bogdan-Ioan Popa (Dept. of Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Ultrasound sensors currently used by autonomous vehicles have short ranges and low resolutions, which limit their uses to proximity sensing. Metamaterial lenses promise to extend the range significantly due to the lenses' excellent beamforming ability, but designs reported so far either have fixed acoustic properties and thus are not able to easily ensonify in desired directions or employ arrays of transducers that are non-scalable and have high power consumption. In this presentation, we introduce a new type

of ultrasound sensor composed of a single transducer and a reconfigurable metamaterial lens operating in air. The metamaterial lens requires less than 0.5 mm movements of a carefully designed surface to form and steer acoustic beams in real-time. The metamaterial tunes all the unit cells simultaneously with only one moving element. Remarkably, steering angles of $\pm 20$ deg are achieved despite the small actuation. The ability to control the ultrasound produced by a single transducer with small actuation is demonstrated experimentally.

**3:35**

**1pSAb3. Passively achieving mode coalescence in acoustic waveguides with simple resonators.** Matthew Kelsten (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, mjk308@scarletmail.rutgers.edu) and Andrew N. Norris (Mech. and Aerosp. Eng., Rutgers Univ., Piscataway, NJ)

A double root, commonly referred to as an exceptional point (EP), for modal frequencies in a 2-D or 3-D waveguide can exhibit almost perfect absorption over a relatively 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 a novel approach to feasibly achieve the aforementioned wall impedance with the use of simple resonators, which can be shown to exhibit mode coalescence at distinct frequencies when treated as a unit cell component of a larger metasurface. In order to evaluate each unit cell design, an efficiency parameter, EP density, was created which quantifies the number of double root frequencies within a given range all while satisfying the following constraints: realizable resonator dimensions, scale separation, and target absorption/attenuation goals. Strategies capable of deriving the necessary resonator parameters for this effect are given. The impact of these strategies on feasibility and performance are discussed via simulated and numerical results. An experiment design consisting of additively manufactured Helmholtz resonators incorporated onto a custom-made impedance tube to simulate the resonant metasurface is discussed along with preliminary results. [Work supported by NSF.]

**3:55**

**1pSAb4. Acoustic focusing using efficient ultrasonic metagratings.** Yan Kei Chiang (School of Eng. and Information Technol., Univ. of New South Wales, Canberra, Campbell, Canberra, Australian Capital Territory 2612, Australia, y.chiang@unsw.edu.au), Li Quan (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX), Yugui Peng, Andrea Alu (Photonics Initiative, Adv. Sci. Res. Ctr., City Univ. of New York, New York, NY), and David Powell (School of Eng. and Information Technol., Univ. of New South Wales, Canberra, Canberra, Australian Capital Territory, Australia)

Metagratings are periodic arrays of rationally designed elements with engineered scattering properties. In contrast to gradient meta-surfaces, a much less dense array of elements is required, and efficient refraction or reflection is possible for very large angles. Previous work at microwave frequencies has shown that they can enhance the performance of lenses with large numerical aperture, by creating a hybrid structure combining metagratings with gradient metasurfaces. However, such structures still require the fine discretization of a gradient metasurface. In this work, we demonstrate that a metagrating can act as a lens by itself, enabling scalable metalenses operating at ultrasonic frequencies. We introduce a new design principle of acoustic metalens by combining multiple metagratings, which enables us to spatially control the reflection angle along the metalens, and hence, to achieve acoustic focusing. The beam steering performance of our proposed metagrating is optimized by a genetic algorithm. A numerical model is established to study the performance of the metalens. Results demonstrate that our metalens can effectively focus the acoustic plane wave to a focal point at ultrasonic range without requiring fine discretization.

MONDAY NOON, 7 DECEMBER 2020 — 12:00 P.M. TO 12:45 P.M.

## Session 1pSCa

## Speech Communication: Memory and Learning in Speech (Poster Session)

All authors will be at their posters from 12:00 noon to 12:45 p.m.

### *Contributed Papers*

**1pSCa1. The effect of virtual reality environments on auditory memory.** Arian Shamei (Linguist, UBC, 2613 West Mal, Vancouver, BC V6T 1Z4, Canada, arianshamei@gmail.com) and Bryan Gick (Linguist, UBC, Vancouver, BC, Canada)

Auditory recall is stronger in the environment in which a memory was originally encoded, an effect of context-dependent memory (CM) [Godden and Baddeley, *Brit. J. Psychol.66*(3), 325–331 (1975)]. Innovations in virtual reality (VR) have resulted in the adoption of VR as a communication platform in professional, medical, and educational contexts. The present study reports an experiment testing how CM impacts auditory memory across differing VR environments. An experiment will be described in which participants in one of two distinct virtual environments (e.g., beach and forest) within the VR-communication platform AltSpace hear three iterations of a pre-recorded list of 16 words controlled for frequency and syllable count. Participants are tested for recall of the word-list in either the same or the differing virtual environment. Improved accuracy when tested in the same environment would suggest that CM can be observed for auditory memory between virtual environments. Preliminary results indicate a potential context-dependent effect between virtual environments. Results will be discussed, with implications for professional, medical, and pedagogical applications in virtual settings.

**1pSCa2. Enhanced memory for sentences read aloud conversationally versus clearly: Evidence from sentence recognition memory and recall.** Sandie Keerstock (Linguist, The Univ. of Texas at Austin, 305 E. 23rd St. CLA 4.400 E9 Mail Code: B5100, Austin, TX 78712, keerstock@utexas.edu), Frida Ballard, and Rajka Smiljanic (Linguist, The Univ. of Texas at Austin, Austin, TX)

Native and non-native listeners were more accurate in identifying sentences as previously heard and in recalling words and entire sentences when they heard clear speech compared to conversational speech (Keerstock and Smiljanic, 2018, 2019). This clear speech benefit on listeners' memory might in part be due to decreased listening effort in perceiving intelligibility-enhancing clear speech ("effortfulness hypothesis," Rabbitt, 1968). The effect of reading sentences aloud in clear speech on *talkers'* memory, however, is unknown. We hypothesized that the effort required to produce clear speech could detrimentally affect memory encoding. In the present study, native and non-native English speakers read sentences aloud in clear and conversational speaking styles. Their memory of the read sentences was assessed either via a sentence recognition memory task (Experiment 1; n = 90) or a recall task (Experiment 2; n = 75). Results from both experiments showed superior retention of spoken information for sentences read aloud conversationally rather than clearly. The results indicate that producing clear speech, unlike perceiving it, interferes with sentence recognition memory and recall. Production of listener-oriented hyper-articulated speech may increase cognitive load leaving fewer cognitive resources available for storing spoken information in memory.

**1pSCa3. Adaptation occurs following a single exposure to altered auditory feedback in speech.** Lana Hantzsch (Waisman Ctr., Univ. of Wisconsin - Madison, 1500 Highland Ave., Rm. 483, Madison, WI 53705, lhantzsch@wisc.edu), Benjamin Parrell, and Caroline A. Niziolek (Commun. Sci. and Disord., Univ. of Wisconsin–Madison, Madison, WI)

Auditory perturbations, or sensory errors caused by manipulating auditory feedback in near real-time, are known to induce instantaneous compensation as well as long-term adaptation responses in speech production. Adaptive responses are built through repetitive exposure to perturbations, but it is unknown how much learning occurs from a single exposure to an auditory perturbation. This trial-by-trial learning, known as one-shot learning, may provide insight into how individual events affect feed-forward control mechanisms and contribute to adaptation. One-shot learning is evident in reaching motor tasks, and the present study aims to understand its role in the speech sensorimotor system. Data were compiled from a set of studies which randomly exposed participants to isolated auditory perturbation events. On any given trial, the participants' first formant (F1) was either shifted up, shifted down, or remained unshifted while they pronounced a presented word. Data from unshifted trials that were preceded by a perturbation trial were extracted to examine if there was a learning effect that carried over from the previous, shifted trial. During unshifted trials that were preceded by a perturbation, participants adjusted their first formant in the opposite direction of the preceding shift, demonstrating that learning occurs even after a single auditory perturbation.

**1pSCa4. Phonetic detail in adult word learning.** Lisa Cox (Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, lmcox@u.northwestern.edu) and Matthew Goldrick (Linguist, Northwestern Univ., Evanston, IL)

What level of phonetic detail do learners use during early stages of word learning? At 14 months, infants taught the name "bih" for a novel object will later accept the mispronunciation "dih" as correct, even though other tasks show infants can reliably hear the difference between these syllables and can successfully complete the same task when the two syllables are less similar (e.g., "lif" and "neem"; Stager and Werker, 1997). White *et al.* (2013) found that when adults hear only a few tokens of a novel word, they act like infants and treat mispronunciations the same as correct pronunciations. We conducted a pre-registered (https://osf.io/7vxpd) partial replication of White *et al*. Adult speakers of English were taught English non-words as labels for images, with 1, 5, or 8 training trials per word. At test, participants heard either a word familiar from training, a single-feature mispronunciation of that word, or a novel word. They simultaneously saw two images on the screen, one familiar from training and one new, and selected the image that best matched an auditorily presented non-word. If White *et al.* successfully replicates, at 1 training exposure participants should treat mispronunciations like correct labels.

**1pSCa5. A second chance for a first impression: Lexically guided perceptual learning reflects cumulative experience with a talker's phonetic input.** Christina Y. Tzeng (Emory Univ., Atlanta, GA), Lynne C. Nygaard (Dept. of Psych., Emory Univ., 36 Eagle Row, Atlanta, GA 30322, lnygaar@emory.edu), and Rachel M. Theodore (Univ. of Connecticut, Storrs, CT)

Listeners use lexical knowledge to modify the mapping from acoustics to speech sounds, but the timecourse of experience that informs lexically guided perceptual learning (LGPL) is unknown. Some data suggest that learning depends on initial exposure to atypical productions, while other data suggest that learning reflects only the most recent exposure. Here we seek to reconcile these findings by assessing the type and timecourse of exposure that promote robust LGPL. In experiment 1, listeners were exposed to ambiguous fricatives embedded in either /s/- or /ʃ/-biasing contexts and then categorized items from an *ashi-asi* continuum at test. Learning was observed; the /s/-bias group showed more *asi* responses compared to the /ʃ/-bias group. In experiment 2, listeners heard both clear and ambiguous productions. Order and lexical bias were manipulated between-subjects, and perceptual learning occurred regardless of the order in which the productions were heard. In experiment 3, listeners heard both /s/- and /ʃ/-biased productions. Order differed between two exposure groups, and no difference between groups was observed at test. Moreover, the results showed a monotonic decrease in learning across experiments, in line with decreasing exposure to stable lexically biasing contexts, and were replicated across novel stimulus sets. The current results did not replicate previous findings showing that either initial or most recent experience are critical for LGPL. Rather, LGPL appears to reflect an aggregation of a talker's input over time.

**1pSCa6. The connection between executive function and phonetic and phonological learning in monolingual and bilingual speakers.** Laura Spinu (Communications & Performing Arts, City Univ. of New York - Kingsborough Community College, Brooklyn, NY, laura.spinu@kbcc.cuny.edu) and Laura Muscalu (Psych., Ithaca College, Ithaca, NY)

Recent work shows bilingual speakers exhibit an advantage in phonetic and phonological learning (PPL) compared to monolinguals. Specifically, bilinguals displayed stronger subcortical encoding of sound when processing speech stimuli (Krizman *et al*., 2012), and outperformed monolinguals in speech perception tasks (Tremblay and Sabourin, 2012). Bilinguals also displayed an advantage in learning vocabularies that differentiated words using foreign phonetic contrasts (Antoniou *et al*., 2015), and various aspects of novel accent pronunciation with both natural and artificial accents (Spinu *et al*., 2018, 2020). The current study explores the mechanisms underlying these enhanced skills by aiming to establish whether a correlation exists between PPL and higher-order cognitive abilities (i.e., executive functions) already shown to be enhanced in bilinguals: attentional control and inhibition (Bialystok, 2017). Our experiment explores the learning of two new segmental patterns in an artificial accent of English (following Spinu *et al*., 2020) in 20 monolingual English speakers and 20 early Spanish-English bilinguals from NYC. We also compare participants' accent learning scores with performance on two classic tasks that assess executive functioning (EF). By revealing a connection between PPL and EF, we add to the body of work on PPL in general, and the bilingual advantage in particular.

**1pSCa7. A gamified Motherese-based method for teaching the expression of emotion in English to speakers of tonal languages.** Natalie Mosseri (Brooklyn College, 3504 Ave. P, Brooklyn, NY 11234, mosserinatalie@gmail.com)

Tonal languages employ different F0 contours to produce linguistic contrasts. Studies suggest native tone language speakers use a more restricted F0 range for expressing emotions in English (Annoli *et al*., 2008, Chong *et al*., 2015). These differences can create communication difficulties

between foreign and native speakers, obscuring the paralinguistic content of their messages. However, intonation training, specifically the expression of emotion, are rarely practiced in ESL classrooms. We test a novel method of teaching intonation without direct native speaker instruction by having learners imitate exaggerated (Motherese-like) sad, happy, and angry sentences, in a gamified format presenting them with appropriate smiley faces. Following training, the participants were tested by reading new sentences in the absence of audio prompts. We included a non-exaggerated version of the stimuli to serve as a control group. The study is currently underway, with 12 ESL learners already tested (target sample size = 24). We scored the non-native productions perceptually, assessing accuracy (the match between perceived emotion and intended emotion). We also visually compared the participants' pitch tracks with those of the native model speech. Both measures converge in revealing the Motherese-based method results in increased accuracy of the various emotions, compared to the method using natural recordings.

**1pSCa8. An analysis of the relationship between linguistic performance and general cognitive abilities in bilingual speakers.** Laura M. Muscalu (Dept. of Psych.,, Ithaca College, Ithaca, NY 14850, muscalu.laura@gmail.com) and Laura Spinu (Communications & Performing Arts, City Univ. of New York - Kingsborough Community College, Brooklyn, NY)

A major issue in bilingualism research is how the bilingual mind processes and reconciles two languages without apparent difficulty when trying to use only one. Research has revealed that when bilinguals produce speech, cross-linguistic competition induced by simultaneous activation of both languages is circumvented by inhibitory and selection mechanisms that allow for phonological articulation of the desired language. Such language-specific processes are presumed to derive from domain-general mechanisms that monitor and regulate behavior, to ensure high levels of cognitive performance. To examine the connection between language-specific and domain-general processes, performance on linguistic and nonlinguistic tasks are compared. In our study, Spanish-English participants perform linguistic tasks that induce competition, inhibition, and selection (e.g., production of translation equivalents that are phonologically similar in the two languages) as well as non-linguistic tasks that induce similar challenges (e.g., alternating between visual or auditory stimuli depending on the task requirements). A fine-grained acoustic analysis of timing and accuracy in verbal production (i.e., VOT, vowel formants, frication spectral peak) will be performed. Positive correlations between performances on the two task categories would support the theoretical view that efficient mental manipulation of two languages relies on control mechanisms that are generally involved in higher-order cognitive behavior.

**1pSCa9. Reading fluency in mono- and multilinguals: Is there an additive effect of the number of languages spoken?** Beckie D. Dugaillard (Communications & Performing Arts, City Univ. of New York - Lehman College, 2001 Oriental Blvd, Brooklyn, NY 11235, beckie.dugaillard@lc.cuny.edu)

Reading is an adequate measure of fluency since it eliminates the use of memory (Pasquarella *et al.*, 2014). Experimental data with lexical access tasks shows reduced fluency in bilinguals compared to monolinguals (Ouzia and Folke, 2016; Sandoval *et al.*, 2010). However, bilinguals also displayed cognitive advantages compared to monolinguals (Bialystok *et al.*, 2012; Spinu *et al.*, 2018, 2020), but these findings are disputed (Marzecova, 2016). The main research aims for this study are to determine whether (1) being a bi- or multilingual is associated with reduced reading fluency and (2) there is an additive effect of the number of languages spoken on reading fluency. Audio recordings of a short English paragraph were collected from 147 undergraduate students of diverse linguistic backgrounds (i.e., monolinguals, bilinguals, and trilinguals). The recordings are currently being analyzed manually for disfluencies such as hesitations, repetitions, mispronunciations / self-corrections, reading speed, and filled pauses (e.g., 'er', 'um', 'ah', 'like'). A final fluency score will be computed for each participant in order to evaluate our research questions. Our study thus expands bilingual fluency research to a reading task, adding to the body of work on multilingual cognition and the controversial phenomenon known as the bilingual advantage.

MONDAY AFTERNOON, 7 DECEMBER 2020 — 1:50 P.M. TO 2:35 P.M.

## Session 1pSCb

## Speech Communication: Language Acquisition and Development (Poster Session)

Authors will be at their posters from 1:50 p.m. to 2:35 p.m.

### *Contributed Papers*

**1pSCb1. The impact of child age and sentence-type on the acoustics of infant-directed speech.** Susan Geffen (Univ. of La Verne, 420 South Madison Ave., 201, Pasadena, CA 91101, segeffen@gmail.com) and Tianlin Wang (Univ. at Albany, Albany, NY)

Research has demonstrated differences in the characteristics of infant-directed speech (IDS) based on age (Stern *et al.*, 1983) and sentence-type (Geffen and Mintz, 2017) but has not examined the two factors together. The current study evaluates whether the acoustics of IDS differ as a function of child's age and sentence-type. The study combines two corpora of native-English adult speakers. Both Corpus1(9mo) (described in Geffen and Mintz, 2017), from the Brent corpus of the CHILDES database (Brent and Siskind, 2001) and Corpus2 (12mo) (described in Thompson, 2019), from naturalistic home-recordings, included statements, yes/no and wh-questions, and were acoustically coded in Praat. Three 2-way mixed ANOVAs with Age (9 and 12 months; between-subjects factor) and Sentence-Type (S, WH, YN; within-subjects factor) on OverallF0range, FinalVowelDuration, and FinalVowelF0range found main effects of Age, and Age X Sentence-Type interaction for OverallF0range. There was also a significant effect of Age on FinalVowelDuration. Results demonstrated a developmental shift in acoustic characteristics of IDS, with more exaggerated prosody to younger

infants, supporting Stern *et al*. (1983) and suggests that IDS to older children no longer privileges prosody as strongly. Future studies should investigate whether similar developmental adjustments in IDS occur in languages other than English.

**1pSCb2. The impact of question prosody during parent-child interactions in a museum setting.** Jill C. Thorson (Commun. Sci. and Disord., Univ. of New Hampshire, 4 Library Way, Hewitt Hall, Dover, NH 03824, jill.thorson@unh.edu), Jill M. Trumbell, and Kimberly Nesbitt (Human Development and Family Studies, Univ. of New Hampshire, Durham, NH)

Children museums provide a dynamic environment for families to engage in learning, with exhibits designed to enhance and stimulate caregiver-child interactions (Marcus, 2016; Willard *et al*., 2019). Additionally, it has been shown that the use of specific types of questions supports narrative development and critical thinking by scaffolding more elaborative responses (Reese and Fivush, 1993). The motivation for this study is to analyze the prosody of *wh*-questions during caregiver-child interactions in an ecologically valid settingã a children's museumã with the goal of better understanding how the prosody of questions may impact language during learning-based free play. Specifically, we examine *wh*-questions during 20-min exhibit explorations by 3- to 6-year old children with their caregivers (n = 10). Caregiver and child *wh*-question prosody is analyzed phonologically (pitch accents and boundary tones) and acoustically (F0). Outcome measures additionally include the overall amount and types of child language produced. We hypothesize that caregiver prosody will be reflected in that of the child, with a correlation between more expressive caregiver prosody and the overall amount of language produced by the child. The broader research aim is to provide strategies to caregivers to increase language output and enhance learning during museum experiences.

**1pSCb3. Developmental changes and effects of parental interaction on French-English bilingual infants' vocalization rates.** Katherine Xu (School of Commun. Sci. & Disord., McGill Univ., 2001 McGill College Ave., 8th Fl., Montreal, QC H3A 1G1, Canada, tian.y.xu@mail.mcgill.ca), Adriel J. Orena (Psych., Univ. of BC, Vancouver, QC, Canada), Yufang Ruan, and Linda Polka (School of Commun. Sci. & Disord., McGill Univ., Montreal, QC, Canada)

In prior work, we investigated how several factors—social context (Social versus Non-Social & One versus Multiple speakers), speaker context (Mother versus Father), language (English versus French) and language dominance (Dominant versus Non-Dominant) – are related to vocalization rates in 10-month-olds growing up in English/French bilingual families (n = 21). This was accomplished by analyzing naturalistic daylong recordings obtained using the Language Environment Analysis (LENA) system. Here, we examined how these factors are related to vocalization rates in older infants by analyzing LENA recordings obtained when these same infants were 18 months of age (n = 16). Similar to previous findings, preliminary analysis showed a higher proportion of infant vocalizations occurred in social contexts (i.e., presence of adult speech). Moreover, more infant vocalizations occurred in contexts with one speaker as opposed to multiple speakers. In contrast to previous findings, 18-month-olds did not vocalize more when interacting with their mothers compared to their fathers. Infant vocalization rates were comparable in French and English input contexts. However, more infant vocalizations occurred in dominant language than in non-dominant language contexts. These findings will help us understand how parent-infant interactions change over time and shape the vocal behavioral of infants being raised in bilingual families.

**1pSCb4. Language input and volubility in French-English bilingual infants.** Yufang Ruan (School of Commun. Sci. & Disord., McGill Univ., 2001 McGill College Ave., 8th Fl., Montreal, QC H3A 1G1, Canada, yufang.ruan@mail.mcgill.ca), Adriel J. Orena (Psych., Univ. of BC, Vancouver, QC, Canada), Katherine Xu, and Linda Polka (School of Commun. Sci. & Disord., McGill Univ., Montreal, QC, Canada)

Caregiver-child interaction is essential for language development. Previous research shows that the amount of one-on-one parentese experienced by monolingual infants predicts their concurrent vocalization (Ramírez-Esparza *et al*., 2014). Here we examined how language and social contexts are related to infant vocalization rates in French-English bilingual infants. Using the Language Environment Analysis (LENA) system, we collected daylong recordings from infants when they were 10- and 18-months of age (N = 21 and 16, respectively). LENA software provided estimations of infant vocalization and adult word counts in 30-second segments. We manually coded half of all segments for *speaker context* (who was speaking around the child), *listener context* (who the speech was directed towards), and *language context* (what language(s) was being used). Three main conclusions emerge from the analyses: (1) the more speech infants hear, the more they vocalize; (2) the input experienced in a one-on-one social context and in the dominant language has a strong relation with infant concurrent and projected volubility; (3) using both raw and proportional language measures to examine bilingual input is recommended. The findings inform our understanding of the relationship between language input and language development in bilingual infants.



**1pSCb5. A comparison of the production of phonetic variants of /t/ in child-directed speech versus adult-directed speech.** Robin Fritche (Linguist, Univ. of Wisconsin-Milwaukee, P.O. Box 413, Milwaukee, WI 53201-0413, rfritche@uwm.edu), Stefanie Shattuck-Hufnagel (Speech Commun. Group, Res. Lab. of Electronics, Massachusetts Inst. of Technol., Cambridge, MA), and Jae Yung Song (Linguist, Univ. of Wisconsin-Milwaukee, Milwaukee, WI)

The surface phonetic details of an utterance affect how 'native' a speaker sounds. However, studies have shown that children's acquisition of context-appropriate variation (sometimes called allophones) is late. This study's goal was to understand how caregivers use phonetic variation in the production of American English /t/ in child-directed speech (CDS), compared to in adult-directed speech (ADS). We hypothesized that mothers modify their input to children in order to produce more limited variation in CDS than in ADS, to potentially assist children in the development of contrastive phonemic categories. To this end, we recorded eight mothers of children under the age of 2 years in both ADS and CDS conditions. Results reveal that CDS contains significantly more canonical cues to /t/ than ADS does, and fewer non-canonical cue patterns, including fewer unreleased tokens in utterance-finally and fewer glottalized tokens both utterance-medially and utterance-finally. Also, in utterance-medial position we found larger aspiration duration differences in CDS between aspirated singleton /t/ vs. unaspirated /t/ in /st/ contexts, suggesting that mothers exaggerate this cue to the phonemic context in which the /t/ occurs. Overall, the finding of more limited variation in CDS may help explain why children produce canonical forms before producing phonetic variants.

**1pSCb6. Emotional speech prosody perception in the first year of life.** Chieh Kao (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, kaoxx096@umn.edu) and Yang Zhang (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Prosodic aspects of speech are considered more salient than linguistic features in prenatal listening environment. The literature on early prosodic speech perception has been largely focused on the role of infant-directed speech in language development. While emotional prosody is also ubiquitous and contains diverse social cues, its perceptual development trajectory remains unclear. The current study adopted the central fixation paradigm to investigate 3- to 12-month-old infants' preferential listening attention to English monosyllabic words spoken in neutral, happy, angry, and sad tones. Forty-six infants completed the experiment and were assigned to younger and older groups with the median cut-off age of 8 months. Linear mixed-effect model results revealed a prevailing trend of shorter listening time for the sad tone. Furthermore, this reduced level of attention was only statistically significant in the younger infants but not the older group, suggesting that the lower-pitched and longer sad prosody was least attractive to the younger infants. With more exposure to emotional speech and the ongoing neural commitment towards language-specific speech perception in the older infants, the early differential interests in voice emotions showed signs of mitigation in the context of spoken words, leading to a relative increase in their listening time to the sad tone.

**1pSCb7. Influence of prosody on acquistion of anticipatory coarticulation in Italian-speaking children.** Patrizia Bonaventura (Speech-Lang. and Hearing Sci., Hofstra Univ., 1000 Hempstead Turnpike, Hempstead, NY 11549, patrizia.bonaventura@hofstra.edu)

The aim of this study was to analyze the influence of prosody on acquisition of temporal aspects of anticipatory lingual coarticulation in productions by Italian-speaking children in a repetition task. Two twin 7-years old male children, native Italian-speakers, interacted with the same adult, repeating disyllables containing VtV sequences where $V_1 = \{i, a\}$ and $V_2 = \{a,e,i,o,u\}$, with different stress patterns (e.g., pi'ta, pita'). The durations of the VC transitions in the different $V_2$ contexts and stress conditions were measured by a spectrographic analysis, and compared between pronunciations by each child versus the adult, to test whether the child was able to imitate the duration of the transitions as produced by the adult in different stress conditions. The results indicated a significant difference in the durations of VC transitions, between the pronunciation of each child and those of the adult they interacted with: longer VC transitions durations were found for one child only in stressed position, but for the other child in both stress conditions. The data seem to support the hypothesis of presence of different temporal aspects of anticipatory coarticulation between adults and children, and of more variability in the children's coarticulatory productions.

**1pSCb8. Development of discrimination and categorization of voice gender cues in school-age children.** Leanne Nagels (Ctr. for Lang. and Cognition Groningen, Univ. of Groningen, University of Groningen, Harmonie Bldg., Oude Kijk in Het Jatstraat 26, Groningen 9712 EK, Netherlands, leanne.nagels@rug.nl), Etienne Gaudrain (CNRS UMR5292, Lyon Neurosci. Res. Ctr., Inserm U1028, Université de Lyon, Lyon, France), Debi Vickers (Cambridge Hearing Group, Sound Lab, Clinical Neurosciences Dept., Univ. of Cambridge, Cambridge, United Kingdom), Petra Hendriks (Ctr. for Lang. and Cognition Groningen, Univ. of Groningen, Groningen, The Netherlands), and Deniz Baékent (Dept. of Otorhinolaryngology/Head and Neck Surgery, Univ. Medical Ctr. Groningen, Univ. of Groningen, Groningen, The Netherlands)

Children's sensitivity to differences in speakers' voices continues to develop throughout childhood, but it remains unclear how it develops. In a cross-sectional design, we investigated how school-age children's discrimination, a lower-level perceptual task, and categorization, a higher-level cognitive task, of voice gender cues, fundamental frequency (F0) and vocal-tract length (VTL), develop and how these abilities are related. We used an adaptive 3AFC procedure to measure children's F0 and VTL discrimination thresholds and a categorization task to examine their weighting of F0 and VTL for voice gender categorization. Here we show that the acquisition of adult-like performance for discrimination and categorization differs across voice gender cues. Children's discrimination thresholds were adult-like around 8 years of age for VTL but still differed from adults' at 12 years of age for F0. Contrarily, children's cue weights were adult-like around 6 years of age for F0 but around 10 years of age for VTL. Therefore, the two abilities seem to develop at different rates for the different voice cues and do not seem to be closely related. Data from cochlear implant children will also be presented, as for them F0 and VTL discrimination and gender categorization depend on both cognitive development and perceptual limitations.

**1pSCb9. Development of formant frequency distributions in American English-speaking elementary school-aged children: A longitudinal study.** Steven M. Lulich (Speech & Hearing Sci., Indiana Univ., Bloomington, IN) and Sherman D. Charles (Speech & Hearing Sci. and Linguist, Indiana Univ., 1610 S Dorchester Dr. Apt. 49, Bloomington, IN 47401, sdcharle@indiana.edu)

Speech development enters a critical phase in the early elementary years, when learned patterns of phonology and motor control become increasingly entrenched, and developmental speech-language disorders become more difficult to treat. Much attention has been given to early speech development in the preschool years, and a few studies have examined the acoustics of speech development cross-sectionally from Kindergarten to adulthood (see Kent and Vorperian, *J Commun Disorders* **74**, 74–97 (2018)]. To the best of our knowledge, there are no longitudinal acoustic studies of the speech of typically developing children between 5 years and puberty. Although cross-sectional studies provide a general picture of growth, they cannot reveal specific patterns of development and variability. We present acoustic data from an annual, longitudinal study of 8 children between 6 and 10 years of age. All participants are native speakers of Midwest American English. Stimuli included words and sentences from the Goldman-Fristoe Test of Articulation 3rd Edition, as well as a spontaneous speech sample and nonce syllables embedded in a carrier phrase. Frequencies for the first four formants were obtained across the entire stimulus set with Praat, and used to generate formant frequency distributions for each child, in each year.

**1pSCb10. Developmental and regional influences on the production of stop closure voicing in late childhood.** Lian J. Arzbecker (Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., 37 Pressey Hall, Columbus, OH 43210, arzbecker.1@osu.edu), Ewa Jacewicz, and Robert A. Fox (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH)

Previous research in sociophonetics examining consonantal variation in American English has uncovered systematic regional differences in the production of voiced stop closures in intersonorant positions, suggesting that social factors can be the source of variation in the phonetic realization of voiced stops. The current study focuses on school-age children to establish both the developmental stability of voiced stops in late childhood and the impact of regional variation on the amount of voicing in stop closures. Based on the literature showing that the mastery of lexical stress contrastivity continues into adolescence, we hypothesized that systematic variation in the amount of stop closure voicing is commensurate with the development of stress control; this relationship is further mediated by regional variation documented in the speech of adults. Sentence productions from 48 girls in the age range 8–13 years representing three regional varieties spoken in Central Ohio, Western North Carolina, and Southeastern Wisconsin were analyzed in a set of temporal variables. Preliminary analyses show moderate correlations between child's age and the variables of interest. Over the course of the development, the relationship between stress contrastivity and the corresponding extent of stop closure voicing is also influenced by regional variation.

**1pSCb11. Preschool aged word learning from storybooks in the presence of background noise.** Madison Buntrock (Hearing & Speech Sci., Univ. of Maryland, 7251 Preinkert Dr., College Park, MD 20742, mbuntroc@terpmail.umd.edu), Emily Shroads, and Rochelle S. Newman (Hearing & Speech Sci., Univ. of Maryland, College Park, MD)

Parents frequently read to their young children, and this can serve as an important opportunity for word learning. While some of these book-reading activities take place in quiet settings, others may occur in the presence of background noise; we were interested in how different types of noise might interfere with word learning. 18 children aged 3.5–5.5 saw and heard a digitized storybook containing 8 consonant-vowel-consonant (CVC) novel "words," and then were tested on their learning. The words were presented in two types of maskers: single male talker in a foreign language and broadband noise. Following the story, word learning was assessed in two ways by: selecting the picture that matched a named referent in a 3AFC and producing the name of a referent. We found a main effect of noise type in the first measure, such that participants had greater accuracy in the broadband noise than in the single talker condition. There was an effect of age in the production measure, but no effect of noise type. These findings are consistent with previous word recognition work that suggests that broadband noise interferes less with learning than a background talker. Future work will expand on the specific features of the maskers that may interfere with word learning success.

## Session 1pSCc

## Speech Communication: Speech Perception (Poster Session)

Authors will be at their posters from 2:50 p.m. to 3:35 p.m.



### *Contributed Papers*

**1pSCc1. Gradient effects and categorical perception co-exist.** Sreeparna Sarkar (Linguist and Cognit. Sci., Univ. of Delaware, 260 South Main St., Newark, DE 19711, sree@udel.edu) and Arild Hestvik (Linguist and Cognit. Sci., Univ. of Delaware, Newark, DE)

Although the perception of speech sounds was originally thought to be purely categorical, subsequent behavioral studies have reported gradient perception. Recent intracranial studies and neuropsychological studies have also shown the existence of both categorical and gradient perception. The current study confirms this co-existence with a simple behavioral task where sensitivity to both within and between-category distinctions are tested within-subject. A VOT continuum of alveolar stops, 0 ms to 100 ms in 5 ms incremental steps of VOT was used to generate 840 pairs of sounds. Participants made a same/different decision for each pair. Results analyzed with Signal Detection Theory show sensitivity to phonetic gradience within-category, and sensitivity increased linearly with greater acoustic distance between the pairs. Interestingly, the linear increase was greater for /d/ than for /t/, which might be attributable to voiceless stops having a more open-ended VOT range. On the other hand, sensitivity to pairs from different categories was significantly greater (as expected), and here, d' did not change by the size of the VOT difference. In conclusion, we confirmed that even though listeners can detect within-category phonetic differences, phonological categorical perception co-exists with phonetic gradient perception.

**1pSCc2. Contributions of acoustic and lexical cue weighting to individual differences in speech perception.** Nikole Giovannone (Univ. of Connecticut, 2 Alethia Dr., Unit 1085, Storrs, CT 06269, nikole.giovannone@uconn.edu) and Rachel M. Theodore (Univ. of Connecticut, Storrs, CT)

Past research suggests that individuals with weaker receptive language show increased reliance on lexical information for speech perception relative to individuals with stronger receptive language, which may reflect a difference in how acoustic and lexical cues are weighted during online speech processing. Here we investigated whether this relationship is the consequence of removing natural correlations between acoustic and lexical cues in speech input, as in the specific stimulus distributions used in previous research. Two groups of adult participants completed measures of receptive and expressive language ability in addition to a measure of lexical recruitment (i.e., a phonetic identification task to measure the Ganong effect). In the high conflict group, the stimulus input distributions put acoustic and lexical cues in high competition; in the low conflict group, this competition was reduced. The results showed that (1) the Ganong effect was larger in the low compared to the high conflict condition, (2) the Ganong effect was larger for those with weaker compared to stronger receptive language, and (3) the relationship between the Ganong effect and receptive language was not mediated by conflict. These results demonstrate increased reliance on lexical information among those with weaker receptive language, suggesting that listeners with weaker language abilities down-weight acoustic cues and rely more heavily on lexical knowledge even when natural acoustic and lexical cue correlations are maintained in the input.

**1pSCc3. Wait, was that *bruise* or *brews*? Utilizing acoustic cues in word recognition of morphologically simple and complex homophones.** Yun J. Kim (Linguist, Emory Univ., Emory University 532 Kilgo Circle 202C, Atlanta, GA 30322-0001, yun.kim@emory.edu)

This study investigates listeners' sensitivity to acoustic cues in recognizing morphologically simple (e.g., *bruise*, *freeze*) and complex (e.g., *brews*, *frees*) homophones. Previous studies demonstrated that native speakers of English use distinct acoustic cues in producing morphologically simple versus complex forms (e.g., Walsh and Parker, 1983; Plag *et al*., 2014; Seyfarth*et al*., 2018). For example, Seyfarth *et al.* (2018) reported that both the stem (e.g., *brew*, *free*) and the segment "si" are longer in duration in morphologically complex words compared to those in monomorphemic counterparts. It still remains in question whether listeners utilize this information in recognizing morphologically simple versus complex homophones. To answer this question, a listening comprehension study is conducted: 20 participants are asked to select the word that they thought they heard. The stimuli have three conditions: (1) both the s and stem length are longer in complex words than simple words, (2) only the s is longer, and (3) only the stem is longer. The influence of each durational cue on the listeners' responses and their response time is analyzed. The results will show whether the distinct acoustic cues available in the input are meaningfully coded and used in speech perception.

**1pSCc4. Towards an understanding of tone category variability in Cantonese.** Rachel Soo (Linguist, Univ. of BC, 100 St. George St., Toronto, ON M5S 3G3, Canada, soorache@gmail.com) and Molly Babel (Linguist, Univ. of BC, Vancouver, BC, Canada)

Cantonese is typically described as having 6 lexical tones. There are reports, however, of three mergers-in-progress; the contrasts between Tone (T) 2 and T5 [Bauer*et al*., *LVC* **15**(2), 211–225 (2003)], T3 and T6, and T4 and T6 [Mok*et al*., *LVC* **25**(3), 341–370 (2013)] are becoming neutralized. Previous work examined the perception of merging tone pairs on an 11-step continuum, in addition to non-merging control pairs (T2-T3 & T5-T6), finding more discrete categorization performance for non-merging than for merging tone pairs. In this study, we examine these same Cantonese tone pair continua with a goodness rating paradigm to understand category-internal structure. Listeners rated the goodness of items on a 7-point scale with real word landmarks as endpoints. Listeners generally rated items across the continua in merging pairs as more acceptable. Items from non-merging tone continua presented a more curious pattern, suggesting that listeners perceived the mid-points of continua as the best-sounding or worst-sounding items. In an attempt to understand these results and those of a previous categorization task, we explore acoustic-auditory similarity measures (e.g., dynamic time warping, Frechet distance, area between curves) between the tones, comparing these measures to listener behaviour.

**1pSCc5. Tonal carryover assimilation is exploited as a speech segmentation cue when cues conflict.** Zhe-chen Guo (Linguist, The Univ. of Texas at Austin, 307 E 31st St. Apt. 105, Austin, TX 78705, y9024131@gmail.com)

Tonal carryover assimilation, whereby a tone is phonetically assimilated to the preceding one, is widely observed across tone languages. This tonal coarticulatory effect is stronger across a smaller prosodic boundary (Lai and Kuang, 2016), suggesting that it may be a speech segmentation cue. The possibility was investigated in two artificial language (AL) learning experiments. Mandarin-speaking participants listened to long utterances in which tokens of the "words" of a three-tone AL (e.g., [pé.tī.kù]) were concatenated without pauses and then identified the words in a test. The first experiment revealed that segmentation was disrupted in an "incongruent-cues" condition where tonal carryover assimilation occurred across AL words in conflict with statistical regularities. Segmentation was neither improved nor inhibited in a "congruent-cues" condition where tonal carryover assimilation occurred within AL words but in only 27% of the word tokens. A follow-up experiment that included a similar congruent-cues condition but maximized the number of cue-bearing word tokens still found a null effect. It is concluded that tonal carryover assimilation is exploited to segment speech in the case of cue incongruence. Yet, it seems redundant when it agrees with statistical regularities, possibly because it is weighted low in the segmentation cue hierarchy (Mattys *et al.*, 2005).

## *Invited Paper*

**2:50**

**1pSCc6. Perceptual analysis on modality change: Is this "Banana" a statement or a yes-no question?** Beatriz O. Câmara (Letras Vernáculas, UFRJ - Universidade Federal do Rio de Janeiro, Cidade Universitária, Rio de Janeiro 21550410, Brazil, beatrizocamara@gmail.com)

This paper attempts to observe what change in the F0 contour is crucial to make Brazilian speakers change their perception of a recording of the word "banana" from a neutral statement to a neutral yes-no question. The Praat software was used to create a close copy stylization ('t Hart 1991) of the word banana,", recorded in the two modalities mentioned before; and, to gradually modify the melodic contour of the statement "banana" until it finally had the same characteristics of the yes-no question contour that was used as a basis. To see which stimuli would change listeners' interpretation of the statment banana to a yes/no question banana, a perception test was made with twenty Brazilian Portuguese native speakers (11 female and 9 male) around the ages of 20 to 60-years-old. The autossegmental-metrical theory by Pierrehumbert (1980) was used to describe Brazilian's intonational patterns for neutral statement and yes-no question.

## *Contributed Papers*

**2:50**

**1pSCc7. Does vowel recognition relate to pitch?** 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), Christian d'Heureuse (Inventec Informatik AG, Winterthur, Switzerland), and Heidy Leemann (Inst. for the Performing Arts and Film, Zurich Univ. of the Arts, Thun, Switzerland)

It was shown that, for vowel sounds, the spectrum relates to fundamental frequency ($f_o$) and the spectral envelope is ambiguous, often representing two or three different vowel qualities if $f_o$ is varied substantially. Thus, from a speech perception perspective, vowel quality is indicated to relate to the pitch of a vowel sound. In this contribution, a concept is outlined addressing the experimental question of the relationship of vowel recognition and pitch. According to this concept, two or three harmonics are extracted near the spectral peaks of a natural vowel sound, representing assumed $F1$–$F2$ or $F1$–$F2$–$F3$, and the lowest harmonic(s) below the first spectral peak are added. Based on a single extracted harmonics pattern, a series of sounds are synthesised stepwise attenuating the levels of the lowest harmonics with the aim to effect a low-to-high transition of the highest common harmonics factor and pitch. Pitch level and vowel recognition of the sounds is then investigated by means of listening tests. Results of a first pilot study based on sounds of mid-closed vowels /e, ø, o/ confirmed the vowel–pitch relation hypothesis and also revealed cases of sounds with double-pitch and double-vowel recognition.

**1pSCc8. Effects of phonetic and indexical variability on talker normalization.** Lee Drown (Univ. of Connecticut, 850 Bolton Rd., Unit 1085, Storrs, CT 06268, lee.drown@uconn.edu) and Rachel M. Theodore (Univ. of Connecticut, Storrs, CT)

Our current work builds on past research demonstrating that listeners experience a processing cost when hearing speech from multiple talkers compared to a single talker. This processing cost is thought to reflect a normalization process during which listeners adjust the mapping to speech sounds to accommodate talker differences in speech production. In the current studies, we use a speeded word identification paradigm to measure processing time for word recognition in single- vs. mixed-talker blocks, and manipulate within-talker and between-talker variability along both phonetic (e.g., vowel formants) and indexical (e.g., fundamental frequency) dimensions. The results to date suggest that listeners incur processing costs given variability in either dimension, even in single-talker blocks, which raises critical methodological considerations for examining talker normalization in addition to informing theories of talker normalization.

**1pSCc9. Effects of filter cutoff on individual speaker identification.** Helen Boyd-Pratt (West Virginia Univ., Morgantown, WV) and Jeremy Donai (Commun. Sci. and Disord., West Virginia Univ., 355 Oakland St., Morgantown, WV 26506, jeremy.donai@mail.wvu.edu)

Two experiments were designed to determine the effects of high-and low-pass filtering on individual speaker identification in quiet and noise. Forty-four listeners ages 18i–35 years of age with normal hearing participated in the two experiments. Each experiment included listening to sentences under various filtered conditions and identifying the speaker. High-and low-pass filter cutoffs included 750, 1500, 2500, and 3500 Hz. Quiet: A significant main effect was found for speaker gender ($F(1, 21) = 6.86$, $p < 0.05$, $\eta p2 = 0.25$), filter cutoff ($F(3, 63) = 4.06$, $p < 0.05$, $\eta p2 = 0.16$), but not filter type ($F(1, 21) = 4.00$, $p$ =0.058, $\eta p2 = 0.16$), which approached statistical significance. Noise: A significant main effect was found for speaker gender ($F(1, 21) = 19.24$, $p < 0.01$, $\eta p2 = 0.48$), filter type ($F(1, 21) = 30.70$, $p < 0.01$, $\eta p2 = 0.59$, and filter cutoff ($F(3, 63) = 6.82$, $p < 0.01$, $\eta p2 = 0.25$). These results suggest that even at the highest high-pass filter cutoffs (i.e., 2500 and 3500 Hz), listeners were capable of extracting information regarding speaker identity and that reducing access to

higher-frequency information had a deleterious effect on identifying the speaker in quiet and noise.

**1pSCc10. Parametrically varying speech adapter length suggests two mechanisms for talker adaptation.** Ja Young Choi (Harvard Univ., 635 Commonwealth Ave., Boston, MA 02215, jayoungc@bu.edu), Rita S. Kou, and Tyler K. Perrachione (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., Boston, MA)

Listeners are slower at processing mixed-talker speech than single-talker speech. This effect of talker variability is reduced when listeners can reorient their attention to each talker via auditory streaming. However, it remains unknown whether sufficient time for attentional reorientation can make speech processing in a mixed-talker context as efficient as processing single-talker speech. Here, we examined how speech processing efficiency for word identification changes with varying lengths of preceding speech. In single- and mixed-talker conditions, listeners identified target words in isolation or preceded by a carrier vowel (/ʌ:/) of parametrically varying durations (300–1500 ms). Word identification was significantly slower in mixed-talker conditions versus single-talker conditions. The response time difference between the mixed- and single-talker conditions significantly decreased as the carrier duration increased from 0 to 600 ms, but carriers longer than 600 ms did not further reduce the additional processing costs in mixed-talker contexts. These results suggest two distinct mechanisms associated with speech processing efficiency: One for rapid adaptation up to 600 ms, reflecting attentional orientation via auditory streaming; and another mechanism operating on longer timescales, perhaps reflecting cognitive resource allocation to accommodate the possibility of talker variability. Our ongoing work explores how these two processes operate over natural speech in mixed-talker contexts.

**1pSCc11. Vowel identification and goodness based on level of formant detail.** Jonathan Jibson (Univ. of Wisconsin–Madison, 7134 Helen C White Hall, 600 N Park St., Madison, WI 53711, jibson@wisc.edu)

Neel (2004) asked how much time-varying formant detail is needed for vowel identification. In that study, starting with vowel recordings from one male speaker and one female speaker, multiple stimuli were synthesized for each vowel: 1-point (monophthongal with midpoint values), 2-point (linear from onset to offset), 3-point, 5-point, and 11-point. Results suggested that a 3-point model was optimal. The present study partially replicates Neel (2004) but draws from more robust phonetic sources (Jacewicz *et al*., 2011; Hillenbrand *et al*., 1995). Eight English monophthongs were chosen for synthesis. 1-, 2-, 3-, and 5-point stimuli were created as described above, and another 1-point stimulus was created with onset values rather than midpoint values. These 40 stimuli were used in two studies ($n = 18$ for each). First was a vowel identification task, where the ten choices were [hVd] words with the chosen vowels (*heed*, *hid*, etc.). Second was a goodness rating task with a 7-point Likert scale, where participants were played a vowel stimulus while being shown the orthographic [hVd] word containing that vowel. The results of neither study showed improvements beyond 2-point stimuli, in contrast to the previous finding of 3-point stimuli as optimal.

**1pSCc12. The perception of English words with consonant clusters and vowel deletion by young normal-hearing listeners under noise.** Eri Iwagami (Sci. and Technolody, Sophia Univ., 7-1 Kioi-cho, Chiyoda-ku, Tokyo 1020083, Japan, alice.erieri@gmail.com) and Takayuki Arai (Information and Commun. Sci., Sophia Univ., Tokyo, Japan)

In this study, we investigated whether the misperception rate of English words with consonant cluster and vowel deletion increase under noise simulated age-related hearing loss. A perception experiment was conducted for young normal-hearing listeners, British English native speakers, in both noise and no-noise conditions. We used real English words were used as target words for this experiment which have consonant clusters in the word-initial and word-final (Type1) and that vowel deletion can occur in first or second vowels (V1 and V2) (Type2). These words were produced and recorded by a male British English native speaker, and the speech sounds were used as the stimuli of this experiment. V1 or V2 in Type2 was produced with deletion and non-deletion. The results showed that the misperception rate of words of Type1 and Type 2 with deletion increased in the noise condition. However, the misperception rate was greater for Type2 with deletion than Type 1 in the noise condition, suggesting vowel energy is important to perceive English words in the noise condition. Hence, the misperception rate by elderly English native listeners is expected to increase for these words as well this result in the simulation of age-related hearing loss.

# Session 1pSCd

## Speech Communication: Neurolinguistics and Psycholinguistics (Poster Session)

Authors will be at their posters from 3:35 p.m. to 4:20 p.m.

### *Contributed Papers*

**1pSCd1. Speech planning impacts word-final consonant variation.** Oriana Kilbourn-Ceron (Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanson, IL 60208, oriana.kilbourn-ceron@northwestern.edu) and Matthew Goldrick (Linguist, Northwestern Univ., Evanston, IL)

We examine how constraints on advance planning in speech production impact word form variation. A /t/ can be produced as a flap only when it's followed by a vowel (e.g., in "atom" and "great exam"). Planning can influence this process by restricting the availability of the following vowel during planning of the /t/. We tested this by examining the reading aloud of adjectives with final /t/ followed by nouns varying in planning difficulty (from hard-to-retrieve low frequency to easily retrieved high frequency; e.g., "great airwave" versus "great exam"). The flap versus non-flap realization of the /t/ was quantified by the percentage of voicing from the offset of the vowel preceding the /t/ target to the onset of the vowel following the /t/. There was less voicing during this interval when the following noun was difficulty to retrieve. This provides new evidence that constraints on speech planning play a role in determining words' phonetic realization, above and beyond the lexical properties of the words themselves.

**1pSCd2. Word-final palatalization and production planning in English.** Donald G. Dunagan (Linguist, Univ. of Georgia, 145 Gilbert Hall, 210 Herty Dr., Athens, GA 30602, donald.dunagan25@uga.edu) and Margaret E. Renwick (Linguist, Univ. of Georgia, Athens, GA)

We analyze palatalization from /z/ to [ʒ] at word boundaries in UK English. Previous research has investigated palatalization in the context of /s#j/, showing that lexical frequency influences its occurrence across word boundaries. Palatalization by voiced coronal fricatives is less well-understood, particularly in naturalistic speech, which we study using the Audio BNC (http://www.phon.ox.ac.uk/AudioBNC). We hypothesize that palatalization across word boundaries is subject to the Production Planning Hypothesis. That is, palatalization should be modulated by the extent to which phonological information for the second word is available when the first word is planned. We analyze spectral center of gravity (CoG) in fricatives from 7,134 word pairs across four phonological contexts, comparing test tokens subject to palatalization, /z#j/ (e.g., "is you"), to control pairs containing non-alternating /z#V/ (e.g., "is it"), or /ʒ/ (e.g., "rouge it," "precision"). Although significant correlates of CoG vary by speaker gender, the acoustics of /z#j/ are predicted by factors related to production planning, including fricative duration, speech rate, presence and length of inter-word pause or sentence boundary, word-pair frequency, and following vowel acoustics. A supervised classification task using acoustics and lexical frequency metrics also distinguishes /z#j/ from /z#V/ and /ʒ#V/ with above-chance accuracy, illustrating automatic detection of palatalization.

**1pSCd3. Using ultrasound imaging to investigate top-down prediction effect on speech motor planning.** Li-Hsin Ning (Dept. of English, National Taiwan Normal Univ., 162, Section 1, Heping E. Rd., Taipei City 106, Taiwan, lihsin@ntnu.edu.tw)

This study investigated whether the activation of speech motor system could be highly influenced by top-down prediction in picture naming when we are listening to prior context. A the picture norming task was administered to 30 native Mandarin speakers (NMSs) to ensure there was an agreement upon the concept names. The pictures (21 out of 31) where the naming accuracy was higher than 0.85 were selected. We then created two stem sentences for each picture/word, one matching the word, and the other one mismatching the word. The 42 tokens (21 words $\times$ 2 stems) were given to another group of 30 NMSs to rate the acceptability. Finally, 12 pairs out of the 21 pairs were selected for ultrasound recording. In the ultrasound recording, the audio signals of stem sentences were played followed by the target pictures displayed on the computer screen. The other group of 30 NMSs had to name the pictures as soon as possible. The results show that articulatory adjustments (lowering the tongue tip) were made at the vowel onset when there was a mismatch between predicted words and to-be-said words. It seems that we react to unexpected words at the early time course of speech motor activation.

**1pSCd4. Neural pitch tracking of nonnative lexical tones in 7 and 11-month-old infants.** Tian Zhao (Inst. for Learning and Brain Sci., Univ. of Washington, Box 367988, Seattle, WA 98195, zhaotc@uw.edu), Fernando Llanos (Dept. of Linguist, Univ. of Texas, Austin, Pittsburgh, PA), Patricia K. Kuhl (Inst. for Learning and Brain Sci., Univ. of Washington, Seattle, WA), and Bharath Chandrasekaran (Dept. of Commun. Sci. and Disord., Univ. of Pittsburgh, Pittsburgh, PA)

Infants' sensitivity to nonnative speech contrasts decline between 6 and 12 months, a period that is considered a 'sensitivity period' for phonetic learning. However, it is yet unknown whether earlier sensory encoding of nonnative speech is also affected by during this key period of transition. To address this question, we recorded frequency-following responses (FFRs) from monolingual English-learning infants at 7 and 11 months of age while they listened to a high-front vowel with a nonnative low-dipping Mandarin lexical tone. The FFR is a scalp-recorded potential that reflects phase-locking activities from cortical and subcortical ensembles in the auditory system. Sixteen infants had at least 1500 clean trials at both ages after data processing. For each age, we assessed the sensory encoding of the nonnative tone by bootstrapping the group-level FFR. We resampled with replacement one thousand16-subject groups. In each iteration, we randomly selected and averaged across 1500 trials of FFR from each subject. The fundamental frequency (*f0*) of each group-averaged FFR was then used to correlate with the *f0* of the stimulus. The results show that the FFR-stim correlation was significantly lower (i.e., worse sensory encoding) at 11 months than at 7 months, suggesting an early sensory marker that may reflect 'perceptual narrowing'.

**1pSCd5. Late positive response indexes neural sensitivity to emotional prosody differences in spoken words.** Chieh Kao (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, kaoxx096@umn.edu) and Yang Zhang (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Emotional prosody is an important social cue that conveys the speaker's intention. Proper recognition of emotional prosody facilitates the interpretation of spoken language. Previous event-related potential (ERP) studies successfully documented an early negative (mismatch negativity, MMN) and a later positive (P3a) involuntary neural responses to detecting a change in

emotional speech prosody (Schirmer *et al*., 2005; Zora *et al*., 2019). Nonetheless, these ERP components were elicited by controlling the linguistic content of the speech stimuli. It remains unclear whether natural affective prosody across varying linguistic carriers would elicit similar activation patterns. The current study adopted the multi-feature oddball paradigm to investigate the ERP responses to three basic emotional prosodiesã happy, angry, and sadã embedded in varying monosyllabic English words. Thirty-three adult listeners (female = 23) completed the experiment. Unlike the previous reports, there was not a clear MMN response to the detection of emotional prosody patterns in the stimuli. But a late positive response (LPR) component was observed frontal-centrally in response to changes in affective prosody. Linear mixed-effect models further confirmed the presence of significantly larger LPRs to happy prosody than angry or sad prosody, suggesting that the LPR is a more reliable neural marker for emotional prosody recognition.

**1pSCd6. Brain-to-brain synchrony in assessing listening effort.** Geoff D. Green (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH), Ewa Jacewicz (Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., 110 Pressey Hall, Columbus, OH 43210, jacewicz.1@osu.edu), Lian J. Arzbecker (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH), Hendrik Santosa (Radiology, Univ. of Pittsburgh, Pittsburgh, PA), and Robert A. Fox (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH)

Evidence from the neuroscience of verbal communication shows that when two people share information (one speaks and the other listens) their brain activities work in synchrony (Silbert *et al*., 2014). This brain-to-brain synchrony (B-Bsync) is lost when the listener fails to understand the speaker's message. We test the hypothesis that B-Bsync predicts the level of effort involved in auditory processing: strong neural brain-to-brain coupling reflects relatively effortless processing; conversely, the weaker the coupling, the greater the effort, and the worse the processing and comprehension. We propose that B-Bsync affords a more sensitive assessment of listening effort than currently available on the basis of behavioral measures. Using functional near-infrared spectroscopy (fNIRS) and fNIRS-based hyperscanning approach, we analyze patterns of neural activity separately in the speaker and in the listener, and assess statistically the correspondence in their brain activation (the degree of synchronized activation of cortical sites and temporal symmetry). We examine the effects of degraded source (i.e., accented American English in speakers telling several short stories) on B-Bsync, predicting the strongest coupling and the shortest time delay when the accent of the listener matches that of the speaker, followed by regional dialect mismatch and foreign-accent mismatch, respectively.

**1pSCd7. Auditory stream discontinuities interfere with speech processing efficiency.** Sung-Joo Lim (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., 610 Commonwealth Ave., Boston, MA 02215-2422, sungjoo@bu.edu), Barbara Shinn-Cunningham (Neurosci. Inst., Carnegie Mellon Univ., Pittsburgh, PA), and Tyler K. Perrachione (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., Boston, MA)

Listeners process speech from mixed talkers less efficiently than from one continuous talker. An auditory streaming framework may help explain this effect: talker discontinuity may disrupt listeners' attentional focus, decreasing speech processing efficiency. In two experiments, we examined how auditory interruption of a speech stream interfered with speech processing. In experiment 1, we sporadically presented non-speech distractors (e.g., text message alerts) while listeners identified words spoken by either one consistent talker or multiple talkers. Word identification was slower immediately following non-speech distractors, and this disruption was greater when listening to speech from a single talker than mixed talkers. Experiment 2 tested the impact of interruptions caused by non-speech distractors or sudden switches to a different talker while listeners identified words spoken by a single, consistent talker. Listeners were slower at responding to non-speech distractors than a new talker's speech, but across interruption types, the magnitude of recovery relative to the magnitude of interruption was similar. These results suggest that both speech and non-speech discontinuities interfere with speech processing by disrupting listeners' attentional focus to a talker. Moreover, interruptions appear less disruptive when listeners pre-deployed cognitive resources to accommodate expected discontinuities, such as when listening to mixed-talker speech.

**1pSCd8. Processing synthetic and human speech in auditory masked priming tasks.** Shiloh Drake (Dept. of Lang., Cultures & Linguist, Bucknell Univ., 1 Dent Dr., Lewisburg, PA 17837, snd006@bucknell.edu) and Erin Liffiton (Dept. of Lang., Cultures & Linguist, Bucknell Univ., Lewisburg, PA)

Over the last decade, synthetic speech has become increasingly realistic and potentially comparable to human speech. Despite it becoming more realistic, do we process synthetic speech in exactly the same way as we do human speech? In this paper, we explore whether humans process synthetic speech in the same way that they process human speech using auditory masked priming (AMP) tasks. AMP operates on the same assumptions that visual masked priming does; namely, that subconscious perception of related words facilitates lexical access. However, results are mixed as to whether synthetic speech is processed similarly to human speech, especially in AMP tasks. For the present experiment, responses were compared across synthetic speech and human speech conditions at three levels (repetition prime, phonologically related prime, unrelated prime) using real words and nonsense words. If processing is similar for synthetic and human speech, the use of synthetic speech to record stimuli for studies will result in a parsimonious and less resource-intensive method to prepare auditory experiments that do not require human-like phonetic contrasts, as well as informing us as to whether we process human and non-human voices similarly.

**1pSCd9. Neural-behavioral relation in phonetic discrimination modulated by language background.** Tian Zhao (Inst. for Learning and Brain Sci., Univ. of Washington, University of Washington, Box 367988, Seattle, WA 98195, zhaotc@uw.edu) and Patricia K. Kuhl (Inst. for Learning and Brain Sci., Univ. of Washington, Seattle, WA)

It is well demonstrated that sensitivities to small acoustic differences in speech contrasts are heavily influenced by the listeners' language background. At the behavioral level, listeners have been shown to discriminate native speech contrasts better than non-native speech contrasts. At the cortical level, mismatch response have been shown to index neural sensitivities to speech contrasts. Using EEG, researchers have observed larger mismatch negativity for native contrasts compared to nonnative contrasts. Using MEG, researchers have described a more focused and efficient neural processing pattern for native contrasts than nonnative contrasts. However, the relation between behavioral discrimination and neural responses are rarely reported. In the current study, 15 native English speakers and 15 native Spanish speakers completed both a behavioral discrimination task and a separate MEG recording using an oddball paradigm. The speech sounds differed on the voice-onset-time (i.e., prevoiced /ba/ with −40 ms VOT versus voiced /ba/ with +10 ms VOT), representing a phonetic contrast only in Spanish. At the group level, English speakers exhibited significantly lower behavioral sensitivity (d') to the contrast and less focused and efficient MEG mismatch responses, replicating previous studies. At the individual level, a significant relation between behavioral sensitivity and the mismatch response only exists in the Spanish group, suggesting potential differences in the mechanisms driving the discrimination for the two groups.

**1pSCd10. How do blind people use semantic and phonological cues for spoken word recognition? An ERP study.** Jie Feng (Beijing Electron. Sci. and Technol. Inst., No.7, Fufeng St., Fengtai Dist., Beijing 100875, China, myonly88@126.com), Chang Liu (The Univ. of Texas at Austin, Austin, TX), Yalan Li (Beijing Electron. Sci. and Technol. Inst., Beijing, China), Peng Sun (Beijing Normal Univ., Beijing, China), Ruibo Xie (Zhejiang Normal Univ., Beijing, China), Ying Zhao, Hongjun Chen, and Xinchun Wu (Beijing Normal Univ., Beijing, China)

Spoken word recognition is affected by phonemic and semantic details of speech at pre-lexical and lexical levels. The current study examined how blind people used semantic and phonological cues for spoken word recognition. Thirty blind and twenty-nine age-matched sighted people participated in this study. We manipulated the semantic similarity and phonological

similarity between the primes and targets in three experimental conditions: semantic-related with different phonology (S+P-), semantic-unrelated with different phonology (S-P-), and semantic-unrelated with the same phonology (S-P+). Results showed that blind participants had higher accuracies than sighted individuals in both the S-P- and S-P+ conditions. As both groups exhibited larger N400 amplitude in the S-P- versus S+P- conditions, the semantic-unrelated N400 effect was stronger in blind than in sighted participants, suggesting a more sensitive processing of semantic information in blind people. Moreover, sighted participants showed stronger N400 effect in the S-P+ than in the S-P- conditions, but blind participants did not, indicating an interference of phonological similarity in spoken word recognition for sighted listeners only. In summary, blind people are more sensitive to semantic cues and less susceptible to phonological similarity interference during spoken word recognition than sighted people.

**1pSCd11. The neurobiological index of perceptual asymmetry in vowel perception.** Megan DiCosta (Commun. Sci. and Disord., St. John's Univ., 8000 Utpoia Parkway, Queens, NY 11439, megan.dicosta17@my.stjohns.edu), Julia Gonzalez (Commun. Sci. and Disord., St. John's Univ., Floral Park, NY), Astero Skliras (St. John's Univ., Kensington, MD), Eghe Adodo (St Johns Univ., Bayside, NY), Valerie Shafer (City Univ. of New York, New York, NY), and Yan H. Yu (Commun. Sci. & Disord., St. John's Univ., Bayside, NY)

Both behavioral and neurophysiological event-related potential (ERP) data has shown that the perception of vowel contrast is often easier in one direction than in the other direction. Several theories and frameworks have increased our understanding of the underlying mechanisms for such asymmetry (e.g., underspecification theory: Eulitz and Lahiri, 2004; the Natural Referent Vowel Framework: Polka and Bohn, 2003, 2011; experience-dependent bias: Kuhl, *et al.* 1991). However, the opposite predictions from these theories call for further studies. The purpose of this study was to examine the neurobiological mechanisms for directional asymmetry in English vowels. We used a mismatch negativity paradigm and presented the English vowel contrast /ɪ/-/ɛ/ (/ɪ/ as in "bit" and /ɛ/ as in "bet") in both directions. The ERP data from 18 young adults were collected. The results showed that the mismatch negativity amplitude was larger when /ɪ/ was used as the standard/frequent/repeated stimulus, and /ɛ/ as the deviant/infrequent/change stimulus. The theoretical implications of such findings will be discussed.

MONDAY AFTERNOON, 7 DECEMBER 2020 — 1:05 P.M. TO 2:35 P.M.

## Session 1pSPa

## Signal Processing in Acoustics: General Topics in Signal Processing in Acoustics I

**Chair's Introduction—1:05**

### *Contributed Papers*

**1:10**

**1pSPa1. Implementing the open master hearing aid on a system-on-chip field programmable gate array.** Ross Snider (Elec. and Comput. Eng., Montana State Univ., 610 Cobleigh Hall, Bozeman, MT 59717, ross.snider@montana.edu), Matthew Blunt, Trevor Vannoy (Elec. and Comput. Eng., Montana State Univ., Bozeman, MT), Dustin Sobrero, Dylan Wickham, and Tyler Davis (Sensor Logic, Bozeman, MT)

System-on-Chip (SoC) Field Programmable Gate Arrays (FPGAs) are ideal for embedded real-time signal processing because of their high performance and low latency. Here we describe the process of implementing the Open Master Hearing Aid [1] on an SoC FPGA. We started by creating a FFT based Simulink model to implement hardware friendly frequency-domain processing. This model implements Short-Time Fourier Transform processing in an overlap-and-add architecture. This was followed by porting additional openMHA processing blocks, such as dynamic range compression, to Simulink. Once the hearing aid Simulink model was finished, Mathwork's HDL Coder was used to create VHDL. To create an interactive system, we used Audio Logic's code generation tools to generate the infrastructure needed to communicate with the hearing aid processor in real-time; this includes generating device drivers that allow Linux to communicate with the hearing aid processor, as well as a custom web application with an autogenerated GUI. This example provides an open reference design for those who may be interested in low latency FPGA based data flow architectures. 1. www.openmha.org

**1:30**

**1pSPa2. Dynamic range compression of sound mixtures.** Ryan M. Corey (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 456 Coordinated Sci. Lab., 1308 West Main St., Urbana, IL 61801, corey1@illinois.edu) and Andrew C. Singer (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Nearly all advanced hearing aids use dynamic range compression (DRC) to make quiet sounds louder and loud sounds quieter. A form of nonlinear gain control, DRC can help to improve listening comfort, but it can also introduce distortion in the presence of multiple competing sounds. The gain applied to all sounds in a mixture is determined by the level of the loudest

sound; the signals therefore modulate each other. It has been widely observed that DRC performs poorly in noisy environments, but there has been little mathematical analysis of the interaction between compression and noise. In this work, we use an idealized model, equating the envelope of a signal with its statistical variance, to analyze the effects of DRC on mixtures of uncorrelated signals. We show that when DRC is applied to a mixture, the effective compression applied to each sound is weaker than it would have been in isolation. Similarly, we analyze how DRC algorithms alter the long-term signal-to-noise ratio of a sound mixture. This analysis can help us to develop DRC algorithms that are more robust to noise and improve the performance of hearing aids in the complex environments where they are needed most.

**1:50**

**1pSPa3. Evaluating the limit of gas phase responsivity with a liquids-coupled ultrasonic tomographic array.** Robert W. Adams (Aramco Res. Center–Houston, Aramco Services Co., 17155 Park Row Dr., Houston, TX 77084, robert.adams@aramcoservices.com), Tao Lin (Aramco Res. Center–Houston, Aramco Services Co., Houston, TX), Muhammad Arsalan (Production Technol. Div., EXPEC Adv. Res. Ctr., Dhahran, Saudi Arabia), and Max Deffenbaugh (Aramco Res. Center–Houston, Aramco Services Co., Houston, TX)

An ultrasonic tomographic array system was developed to measure oil/water/gas phase fraction in multiphase flows. This system interrogates fluids with a ring-array of ultrasonic transducers to generate tomographic images and sound speed profiles of the fluid. Gas entrained within the flow presents a signal transmission challenge for this system, due to reflections at liquid/gas interfaces and absorption of ultrasonic energy. Techniques were developed using physical models and machine learning based algorithms to extend the operating envelope of this system in multiphase fluid flows containing gas entrained in the flow. The array system measures the travel time of the ultrasonic pulses between transceiver pairs. As the signal-to-noise (SNR) at the receiver decreases due to the absorption of the signal due to the presence of gas in the fluid, the variance of the travel time measurement increases. Travel time measurements may also be distorted by reflections at liquid/gas phases leading to longer ray paths between transducer pairs. The upper limit of gas phase measurements is evaluated using physical models for the travel time of pulses between transceiver pairs. Machine learning techniques are applied to determine the gas phase fraction and increase accuracy of the liquids-phase fraction measurements of the system.

**2:10**

**1pSPa4. Modifying the Welch method to estimate power spectral percentiles.** Felix Schwock (Elec. and Comput. Eng., Univ. of Washington, 185 Stevens Way, Paul Allen Ctr. – Rm. AE100R, Seattle, WA 98195, fschwock@uw.edu) and Shima Abadi (Elec. and Comput. Eng., Univ. of Washington, Seattle, WA)

Welch overlap segment averaging method is a popular approach for estimating power spectra of stochastic signals due to its computational efficiency, its ability to scale estimation variance, and its potential to reduce spectral leakage. However, in its original form, the Welch method is prone to spectral outliers caused by transient signals. A computationally efficient solution is to replace the common mean averaging for each frequency bin by a percentile estimation, which has proven to be a robust alternative to the original method. The statistical properties of this approach, such as estimation variance and limiting distribution, have not yet been analyzed in greater detail. In this talk, we present respective expressions for the Welch percentile estimator by using concepts from order statistics and spectral estimation theory. The Welch percentile estimator is applied to the ocean ambient noise data which are compromised by transient signals from an Acoustic Doppler Current Profiler (ADCP) co-located with the hydrophone. Based on the statistical properties of the percentile estimation, confidence intervals for the ocean noise levels have been computed to provide a measure for the estimation quality.

## Session 1pSPb

# Signal Processing in Acoustics: General Topics in Signal Processing in Acoustics II

Michael B. Muehlstein
*Cold Regions Res. and Eng. Lab., U.S. Army Eng. and Res. Development Lab., Hanover, NH 03755*

**Chair's Introduction—2:50**

### *Contributed Papers*

**2:55**

**1pSPb1. Versatile system analysis procedure based on a frequency domain variant of velvet noise.** Hideki Kawahara (Wakayama Univ., 930 Sakaedani, Wakayama 640-8510, Japan, kawahara@wakayama-u.ac.jp), Ken-Ichi Sakakibara (Health Sci. Univ. of Hokkaido, Ishikari-gun, Japan), Mitsunori Mizumachi (Kyushu Inst. of Technol., Kitakyushu, Japan), Masanori Morise, and Hideki Banno (Meijo Univ., Nagoya, Japan)

We introduce a method that enables the simultaneous measurement of system attributes. The attributes are the linear time-invariant, nonlinear time-invariant, and random and extra responses without introducing additional equipment and post-processing. This new procedure uses a new member of time stretched pulses called FVN (Frequency domain variant of Velvet Noise). A high degree of design freedom makes individual FVN generated using different random number seeds plays the key for making orthogonal sequences. We made this procedure and an interactive and realtime acoustic measurement tool in an open-source repository. This procedure and tool enable in-depth measurement of acoustic systems in everyday conditions. The proposed procedure is general and applies to many fields such as altered auditory feedback experiments and assessment of recording and coding effects on voice research.

**3:15**

**1pSPb2. CDMA-based multi-domain communications network for marine robots.** Jay Patel (Elec. and Comput. Eng., Dalhousie Univ., 1360 Barrington St., Halifax, NS B3H 4R2, Canada, patel.jay@dal.ca) and Mae Seto (Elec. and Comput. Eng., Dalhousie Univ., Halifax, NS, Canada)

A cross-domain communications network for above and below water marine robots, based on code-division multiple access (CDMA), is reported. CDMA is a promising physical layer and multiple access technique for underwater acoustic sensor networks as it: (i) is robust to frequency selective fading, (ii) compensates for multi-path effects at the receiver, and (iii) allows receivers to distinguish amongst signals simultaneously transmitted by multiple devices. Consequently, CDMA increases channel re-use and reduces packet retransmissions, which results in decreased energy consumption and increased network throughput. The proposed CDMA network for autonomous co-ordination and networking is applied to marine robots separated by extended ranges to transmit images/information from underwater to above-water. The work involves a complete communications protocol stack from the physical to the application layer. Simulations of the proposed network were performed with Network Simulator-3 (NS-3). The proposed protocol leverages CDMA properties to achieve multiple access to the scarce underwater bandwidth while previous reported work with underwater channels only consider CDMA for the physical layer encoding. Simulations shows the proposed underwater acoustic network protocol outperforms other existing ones. The next step is preliminary testing in-water.

**3:35**

**1pSPb3. High data rate ultrasonic wireless communication through *in vivo* biological tissues and simulations.** Gizem Tabak (Univ. of Illinois at Urbana-Champaign, 1308 W Main St., 119 Coordinated Sci. Lab., Urbana, IL 61801, tabak2@illinois.edu), Michael L. Oelze (ECE, Univ. of Illinois at Urbana-Champaign, Urbana, IL), and Andrew C. Singer (Univ. of Illinois at Urbana-Champaign, Urbana, IL)

The use of wireless implanted medical devices is growing because they reduce the discomfort of patients and the likelihood of infection associated with trailing wires. Currently, radiofrequency electromagnetic waves are the most commonly used method for communicating wirelessly with IMDs. However, due to the restrictions on the available bandwidth and transmitted power, the data rates of RF-based IMDs are limited. Previously, we introduced signal processing and communications methods that use phase-coherent decision feedback equalization to relay information robustly at high data rates using ultrasonic waves. The experiments performed with *ex vivo* biological tissues in a water tank demonstrated data rates greater than 4 Mbps with bit error rate less than 1e-4 using mm-sized, biocompatible transducers. In this study, we first employ the proposed system in simulations conducted with finite impulse response channel models obtained with mm-sized transducers in phantoms and demonstrate video-capable data rates at moderate signal-to-noise ratio levels. We then perform *in situ* and *in vivo* physical experiments with rabbits, and achieve 3.2 and 6.2 Mbps data rates, respectively. Finally, we quantify the degrading effects of acoustic nonlinearities on the data rates by using nonlinear propagation models for biological tissues, and we propose ways to mitigate such effects.

**3:55**

**1pSPb4. Acoustic communication along drill strings for deep subsurface monitoring.** Xiaoqin ZangANG (Energy and Environment Directorate, Pacific Northwest National Lab., PO Box 999 MSIN #K9-33, Richland, WA 99354, xiaoqin.zang@pnnl.gov), Yang Yang, Jayson J. Martinez, Aljon Salalila, Zhiqun Deng, and Christopher E. Strickland (Energy and Environment Directorate, Pacific Northwest National Lab., Richland, WA)

Monitoring at a carbon storage site is necessary to track the movement of $CO_2$ and assure permanence for geologic storage. Therefore, advanced monitoring technologies and effective methods to transmit data from downhole to the surface are needed to decrease the cost and uncertainty in measurements and satisfy regulations for tracking the fate of subsurface $CO_2$. This study covers the design, development, and initial laboratory-scale feasibility study of an acoustic communication system for monitoring the deep subsurface. We set up a metal tubing in lab space as the model and designed the hardware system for acoustic signal transmission and data collection. The attenuation of the signal along the channel and the impulse response of the acoustic channel were characterized. Using an inverse filter processing method, communication signals were successfully recovered and decoded with zero-bit error rates at three testing locations.

## Session 1pID

## Interdisciplinary: Keynote Lecture: A Digital Stethoscope with Active Noise Suppression and Automatic Detection of Abnormalities in Lung Sounds

Derrick Knight, Cochair
*Trane Technologies, 3600 Pammel Creek Road, La Crosse, WI 54601*

Andrew Schmidt, Cochair
*USG Corporation, 700 N Highway 45, Libertyville, IL 60048*



**Chair's Introduction—4:30**

### *Invited Paper*

**4:35**

**1pID1. A digital stethoscope with active noise suppression and automatic detection of abnormalities in lung sounds.** James West (Johns Hopkins Univ., 3400 North Charles St. Barton Hall 105, Baltimore, MD 21218, jimwest@jhu.edu) and Ellington S. West (Sonavi Labs, Baltimore, MD)

Auscultation, the action of listening to sounds from the body, typically with a stethoscope, as a part of medical diagnosis remains one of the most common, and cost-effective diagnostic practices but requires a high level of expertise. Although widely practiced, it is undermined by subjectivity in interpretation, limiting the ability to accurately interpret sounds objectively and repeatedly. Frequently, high environmental noise levels render conventional stethoscopes useless. It is also true that substantial experience is required in order to properly diagnose lung abnormalities such as pneumonia and Covid-19. Here we present a digital stethoscope with active noise suppression and an artificial intelligence algorithm (AI) that identifies lung abnormalities with accuracy comparable to trained medical personal. This new line of respiratory diagnostic tools is appropriate for community health workers in under-resourced regions, for chronic respiratory patients in their home, and for medical professionals in noisy clinics, who wish to improve their ability to hear and interpret lung sounds. This conversation will also explore the opportunities and obstacles associated with bringing a product from the lab to the marketplace.

# Session 2aAAa

## Architectural Acoustics, Noise, and Structural Acoustics and Vibration: Sound Transmission and Impact Noise in Buildings I

Matthew Golden, Cochair
*Pliteq, 131 Royal Group Crecent, Woodbridge, L4H 1X9, Canada*

Benjamin M. Shafer, Cochair
*PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406*

**Chair's Introduction—9:30**

### *Invited Papers*

**9:35**

**2aAAa1. Laser Doppler vibrometry-based characterization of bending waves by rotating viscoelastic panel materials applied in wallboard systems.** Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 Eighth St., Troy, NY 12180, xiangn@rpi.edu) and Max Miller III (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

A low-cost, thin-panel form of viscoelastic materials has found potential applications as constrained damping layers integrated into gypsum wallboards. The standardized chamber-based random-incident tests demonstrate drastically increased sound transmission losses without significantly increasing the board thickness and weight when the viscoelastic thin panels are integrated into gypsum wallboards in sandwich structure. For better understanding of their excellent effect as constrained damping layers, this work applies a laser Doppler vibrometer-based method in characterizing bending wave properties of the viscoelastic panel materials. The method involves a time-domain broadband measurement technique which correlates impulse responses taken at two points radially away from a bending wave source. Using reasonable sample sizes, boundary reflections will still obscure measurement results, particularly towards low frequencies. By rotating the panel under test, the coherence of direct wave pulses measured equidistant from the source and the lack thereof disturbing boundary reflections enables largely removing the undesirable reflections. In this way the direct wave pulses can be extracted by angular averaging and windowing in the time-domain. This paper discusses experimental investigations on the characterization method of bending wave properties using a laser Doppler vibrometer.

**9:55**

**2aAAa2. Measurement and mitigation of heavy-weight impacts.** Wayland Dong (Veneklasen Assoc., 1711 Sixteenth St., Santa Monica, California, CA 90404, wdong@veneklasen.com), John Lo Verde, Samantha Rawlings, and Richard Silva (Veneklasen Assoc., Santa Monica, CA)

Various mitigation systems have been utilized for addressing heavy-weight impacts within the built environment. There is no standardized measurement system for assessing heavy-weight impacts and no metric for describing the performance of a building assembly, material, or sound level for this type of impact. The authors have continued their investigation into measurement methods for hard and soft heavy-weight impacts on a variety of structures and fitness flooring materials. This paper presents recent results.

**10:15**

**2aAAa3. Developing criteria for fitness impacts.** Samantha Rawlings (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, srawlings@veneklasen.com), Wayland Dong, and John Lo Verde (Veneklasen Assoc., Santa Monica, CA)

Heavy-weight impacts are known to cause disturbances in mixed-use buildings. Most attention has been given to mitigating impacts when complaints occur. However, for a designer or a developer attempting to define acceptability prior to construction, there are no guidelines for defining what occupants consider acceptable. This paper follows previous work examining ambient vibration levels in a variety of building types, with and without complaints, to identify potential vibration thresholds at which complaints may occur.

10:35

**2aAAa4. A new test standard for measuring impact noise in buildings.** Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027, jerry@jglacoustics.com)

In 2018, ASTM International published a new laboratory test standard for measuring impact noise in buildings. Unlike previous ASTM standards for impact noise, which assess impact noise transmission into nearyby spaces, this test method evaluates impact sound radiating into the source room. This presentation will review the history of the development of this standard, discuss limitations of the current standard, and suggest improvements that are being considered.

TUESDAY MORNING, 8 DECEMBER 2020 — 11:15 A.M. TO 12:45 P.M.

## Session 2aAAb

## Architectural Acoustics, Noise, and Structural Acoustics and Vibration: Sound Transmission and Impact Noise in Buildings II

Matthew Golden, Cochair
*Pliteq, 131 Royal Group Crecent, Woodbridge, L4H 1X9, Canada*

Benjamin M. Shafer, Cochair
*PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406*

**Chair's Introduction—11:15**

### *Invited Papers*

11:20

**2aAAb1. Measurement of low frequency impact insulation.** Wayland Dong (Veneklasen Assoc., 1711 Sixteenth St., Santa Monica, California, CA 90404, wdong@veneklasen.com), John Lo Verde, and Sunit Girdhar (Veneklasen Assoc., Santa Monica, CA)

The measurement of low-frequency impact sound (below 100 Hz) in rooms is critical to determining human response and acceptability of floor ceilings [LoVerde and Dong, *J. Acoust. Soc. Am.* **141**, 428–440 (2017)]. Measurement uncertainties generally increase at lower frequencies, and a lack of precision is a general problem that ratings of low-frequency noise must overcome. Impact noise uncertainties at low frequencies are reviewed. The effect on measurement uncertainty of changes to the measurement procedure (such as fixed versus roving microphones, measurement duration, microphone position, number of tapping machine locations, etc.) is studied, with emphasis on the uncertainties in the 50–80 Hz third-octave bands. The measurement uncertainty will be related to the precision desired to accurately relate to human reaction.

11:40

**2aAAb2. Evaluation of laboratory low-frequency impact noise data.** Ryan L. Skoug (ESI Eng., 7831 Glenroy Rd., Ste. 430, Minneapolis, MN 55439, rskoug@esi-engineering.com)

Acoustic consultants and manufacturers of sound isolation products are becoming increasingly concerned about low-frequency footfall noise transmission through floor/ceiling assemblies in multi-family buildings. While the typical IIC rating only evaluates impact noise down to the 100 Hz 1/3 octave band frequency, some product manufacturers are beginning to evaluate impact sound isolation down to the 50 Hz, or even 20 Hz frequency. In this presentation, we will review low-frequency impact noise laboratory test results produced with both the standard tapping machine and the Tachibana impact ball, comparing the results to each other, possible criteria, and expected outcomes.

**12:00**

**2aAAb3. Correlation methods of room impulse responses for chamber-based sound insulation measurements.** Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 Eighth St., Troy, NY 12180, xiangn@rpi.edu)

Sound insulation of building elements, such as walls, doors and windows are quantified in standardized test chambers (ISO 10140, 2010; ASTM E90-09, 2016). Standardized test methods employ continuous noise signals to excite the sending room. The sound insulation is determined through averaged sound level differences between the sending and receiving rooms with a correction term taking into account absorption in the receiving room. High insulation elements often post challenges for reliable measurement results since insufficient signal-to-noise ratios in the receiving room make the measurement inaccurate, often invalid. This paper discusses advantages of measurement techniques using pseudo random noise and /or log-sweep sine signals followed by efficient cross-correlation processing, leading to room impulse responses. Energy building of room impulse responses measured at microphone positions in both rooms also yields spatially averaged differences of sound energy levels. In addition, the room impulse responses measured in the receiving rooms also provide reverberation times to evaluate overall room absorption. Due to cross-correlation, the room impulse responses even in low signal-to-noise situations can be measured with drastically improved quality, yielding more reliable evaluation of level differences. This paper highlights theoretic fundamentals of correlation methods, their equivalence, and differences with the classical noise excitation method.

**12:20**

**2aAAb4. Updated comparison of different resilient channels with and without isolators.** Elzo F. Gernhart (Product Development, PAC Int., LLC, 7260 W Azure Dr., Ste. 140-213, Las Vegas, NV 89130, elzo@pac-intl.com)

It is the purpose of this presentation to provide a current update of the major resilient channels available with and without acoustical isolators in floor-ceiling assemblies with and without various finished floor systems. All the testing was conducted with in twelve months of the date of this paper. All the samples where either purchased at retail outlets or donated by the manufacturers for these tests. Here in we compare the most popular brand names and models of resilient channels with and without resilient channel isolators. The finished flooring and finished flooring mats where standardized to reduce the number of finished flooring tests required to demonstrate the advantages of the resilient channels and the resilient isolator. All though this presentation may not include 100% every available product. We have selected the most commonly used products: resilient channels, rubber mat and finished flooring. In open web timber truss, TJI and solid timber joist. This research should be important to every acoustical engineer whom is concerned about delivering current state of the art, cost effective, noise control solutions to the client.

## Session 2aABa

# Animal Bioacoustics and Psychological and Physiological Acoustics: Celebrating Peter Narins' Contributions to Auditory Science I

Mark A. Bee, Cochair
*Ecology, Evolution & Behavior, University of Minnesota, 140 Gortner Laboratories, 1479 Gortner Ave., St. Paul, MN 55108*

Andrea M. Simmons, Cochair
*Cognitive, Linguistic, & Psychological Sciences, Brown University, 190 Thayer St., Box 1821, Providence, RI 02912-9067*

**Chair's Introduction—9:30**



### *Invited Papers*

**9:35**

**2aABa1. From sensory neuroethology to science diplomacy and policy: Unexpected passions and rewards.** John G. Hildebrand (Dept. of Neurosci., Univ. of Arizona, 1050 E. 4th St., Rm. 611, POB 210077, Tucson, AZ 85721-0077, jhildebr@email.arizona.edu)

For 46 years, my research focused on the neurobiology and behavior of arthropodsã particularly insects and crustaceans. During most of that time, my goal was to understand neurobiological mechanisms through which information about behaviorally significant olfactory stimuli is encoded, processed, and integrated with inputs of other modalities in the insect brain and to learn how those natural chemical stimuli trigger and control specific, adaptive behaviors. But I am here not to talk about olfaction in an acoustical sciences forum, but rather to contribute to honoring my long-standing friend and colleague, Peter Narins. I will speak about the work I have been doing since 2014, as the Foreign Secretary of the National Academy of Sciences, one of the five elected officers of the NAS. To put my work in context, I will consider the origins of the NAS and other national science academies; the purposes and activities of the NAS; and its international scope and endeavors. Finally, from my current perspective, I will celebrate certain renowned achievements of our honoree.

**9:55**

**2aABa2. In honor of Peter Narin's retirement: Frogs and morse code operators.** Richard A. Schmiedt (Hearing Res. Program, Otolaryngol. - Head and Neck Surgery, Medical Univ. of South Carolina, 135 Rutledge Ave., MSC#550, Charleston, SC 29425-5500, schmiera@musc.edu)

Acoustic communication between animals runs the gamut of simple to complex in its dynamic time and frequency structure. Narrow bandwidth codes are often used in noisy environments wherein simple filters can be used to recover the signal under poor signal to noise ratios. Tree frogs in the genus *Eleutherodactylus* use essentially single and double tones to attract mates in very noisy environments with many males calling mostly silent females. Over the last 40 years, Peter Narins has been studying these simple frog codes. He has shown that these frogs have evolved behaviors to avoid acoustic interference with their neighbors, both in time (by call suppression) and by changing their dominant frequencies across species. Radio amateurs using Morse code mimic frog behavior when trying to communicate with highly desired, weak-signal DX (distant) stations, where many strong signals are calling simultaneously. In this scenario, termed a "DX Pileup," the calling operators use strategies similar to those of frogs to successfully contact the DX station. Peter (K2IXQ) has been an avid radio amateur for most of his life, using Morse code to chase DX around the world. Clearly, there is an overlap between his scientific research and his personal hobby.

**10:15**

**2aABa3. Peter Narins: From laboratory to field and back again.** H C. Gerhardt (Biological Sci., Univ. of Missouri, 215 Tucker Hall, Columbia, MO 65211, gerhardth@missouri.edu)

Scientists trained as engineers rarely venture into the natural world to study the behavior of animals whose auditory systems they analyze in the laboratory. But Peter Narins began his career as a graduate student studying the behavior of the co-qui frog in Puerto Rico; his goal was to learn the biological significance of the two notes of the calls of this species, whose peripheral auditory system he was studying in Bob Capranica's laboratory at Cornell. He found that the two different inner ear organs – typical of anuran amphibians – were tuned to different frequencies corresponding to those of the two notes and that these notes played different roles in the communication system of this species. Peter continued to delve deeper into the auditory system of this and other frogs, making significant discoveries about traveling waves, two-tone inhibition, otoacoustic emissions, and phase-locking. These discoveries were then related to further behavioral analyses of acoustic pattern recognition and sound localization, which is based on a pressure-difference system. Finally, Peter's field work has included studies of exotic species that produce complex signals, ultrasonic signals, which are audible to them, and sensory systems underlying seismic communication.

## Session 2aABb

# Animal Bioacoustics and Psychological and Physiological Acoustics: Celebrating Peter Narins' Contributions to Auditory Science II

Mark A. Bee, Cochair
*Ecology, Evolution & Behavior, University of Minnesota, 140 Gortner Laboratories, 1479 Gortner Ave., St. Paul, MN 55108*

Andrea M. Simmons, Cochair
*Cognitive, Linguistic, & Psychological Sciences, Brown University, 190 Thayer St., Box 1821, Providence, RI 02912-9067*

**Chair's Introduction—11:15**

### *Invited Papers*

**11:20**

**2aABb1. Nonlinear effects and economically irrational decisions in a frog communication system.** Michael J. Ryan (Dept. of Integrative Biology C0930, Univ. of Texas, 2415 Speedway Ave., Austin, TX 78712-1064, mryan@utexas.edu)

The general match between biases in the auditory system and spectral and temporal properties of the species-specific advertisement call have made anurans an enviable model for studies of the neuroethology of acoustic communication. Professor Narins' earlier studies on sexual dimorphism in coqui communication is a sterling example. We have revealed a similar congruence between peripheral and central auditory biases in the recognition and preference for conspecific mating calls in túngara frogs. Studies of cognitive aspects of mate choice, however, have revealed a number of difficult to predict nonlinear effects, and even economically irrational decisions, in this communication system; examples include: Weber's law, competitive decoy effects, and perceptual rescue in audio-visual communication. I will conclude by discussing preliminary data on how neuropeptides involved in the neural reward circuits can broaden the acoustic recognition landscape of these frogs.

**11:40**

**2aABb2. Unusual phenomena in sound communication and hearing in anuran amphibians.** Albert S. Feng (Molecular and Integrative Physiol., Univ. of Illinois at Urbana-Champaign, 524 Burrill Hall, 407 South Goodwin Ave., Urbana, IL 61801, fengatcu@gmail.com)

The ability of anuran amphibians to detect, recognize and localize sound is important for their reproductive success, mate choice and competitive interactions between calling males. Recent studies have revealed that the view of frogs possessing only a crude perceptual ability is overly simplistic, or wrong altogether. Whereas the evidence of the antithesis is broad, only some of the evidence will be highlighted here due to space limitation. (1) Two species of torrent frogs that inhabit noisy streams characterized by intense but predominantly low-frequency stream noise have been shown to have evolved the ability to hear and communicate with ultrasound. (2) Despite the small body size and short interaural distance of male *Odorrana tomorta*, these animals can localize sound with an acuity that rivals those of barn owls, dolphins and humans. (3) These same animals uniquely possess (among the vertebrates) the ability to actively control the frequency sensitivity of hearing. (4) The vocal signals of these animals show huge intra- and interindividual variability, and despite this they show the ability to discriminate neighbors' and strangers' calls. Although we have some insights into the mechanisms underlying these unusual hearing abilities much work remains to be done for gaining a comprehensive understanding.

**12:00**

**2aABb3. *Allobates femoralis*: An anuran model species for field bioacoustics, behavioural ecology and cognition.** Walter Hödl (Dept. of Evolutionary Biology, Univ. of Vienna, Althanstrasse 14, Wien 1090, Austria, walter.hoedl@univie.ac.at)

The males of the brilliant-thighed poison frog, *Allobates femoralis*, are known to present stereotypic phonotactic responses to the playback of conspecific and synthetic calls. Fixed site attachment and a long calling period render this terrestrial and diurnal pan-Amazonian frog a rewarding species for bioacoustics. Several aspects of field bioacoustics in A. femoralis have been studied by the author's research team during the last 35 years such as the influence of temporal and spectral variation of the advertisement call on phonotactic responses. A clear difference in the phonotactic behaviour was found in populations with or without a co-occuring species presenting an overlap in the frequency range. With the use of a robot frog it could be shown that a combination of acoustic and a visual signal is necessary to elicit agonistic behaviour in this highly territorial species. In *A. femoralis* obligatory tadpole transport is generally performed by males, whereas females abandon their clutches after oviposition. However, when males are removed (and unable to acoustically announce their presence) females flexibly take over their parental duty and carry the tadpoles to aquatic sites. Contrary to the view of amphibian parental care as being stereotyped, these results demonstrate behavioural flexibility as an adaptive response to environmental and social uncertainty. Tracking experiments with territorial and/or tadpole carrying males revealed accurate homing or water finding trajectories suggesting the integration of learned landmarks.

# Session 2aBAa

## Biomedical Acoustics, Signal Processing in Acoustics, Computational Acoustics, and Physical Acoustics: Modelling and Measuring Nonlinear Ultrasound Signals I

Thomas L. Szabo, Cochair
*Biomedical Engineering, Boston University, 44 Cummington Mall, Boston, MA 02215*

Keith A. Wear, Cochair
*Center for Devices and Radiological Health, Food and Drug Administration, Bldg 62 Room 2114, 10903 New Hampshire Ave., Silver Spring, MD 20993*

**Chair's Introduction—9:30**



### *Invited Papers*

**9:35**

**2aBAa1. Addressing the challenges of hydrophone measurements in nonlinear ultrasonic fields.** Andrew M. Hurrell (Precision Acoust. Ltd., Hampton Farm Business Park, Higher Bockhampton, Dorchester, Dorset DT2 8QH, United Kingdom, andrew@acoustics.co.uk)

The use of hydrophones to quantify ultrasonic output of nonlinear fields may be beset with pitfalls for the unsuspecting experimentalist. This paper discusses a variety of issues that need to be considered including, but not limited to: broadband hydrophone frequency response; frequency dependent directivity; risks associated with device overload and/or damage in the context of high amplitude fields; influence of EM interference. Methods to identify such effects will be presented. These will be accompanied by a review of good experimental practice aimed at minimising undesirable measurement artefacts and reducing potential damage to metrological apparatus and instrumentation.

**9:55**

**2aBAa2. An example usage of a reflectance-based fiber-optic hydrophone in a high pressure field.** Sam Howard (Onda Corp., 1299 Hammerwood Ave., Sunnyvale, CA 94089, sh@ondacorp.com)

Building upon recent work on the calibration of reflectance-based fiber-optic hydrophones [1], a measurement example is presented wherein a 1.45 MHz F/1 100mm diameter source is characterized at varying levels of driving voltage, up to free-field acoustic pressure amplitudes of 80.0 MPa (peak-compression) and 13.6 MPa (peak rarefaction) with harmonic content beyond 150 MHz. The method of calibration and correction for nonlinear effects are reviewed and discussed, as are consideration of mechanical effects such as cavitation and fiber-deflection due to radiation force. [1] S. Howard and C. Zanelli, "Validation of reflectance-based fiber-optic hydrophones, *International Congress on Acoustics,* Aachen, Germany, 2019

**10:15**

**2aBAa3. Improved characterization of ultrasound sources by radiation force balance using acoustic holography in place of plane-wave and geometric approximations.** Oleg A. Sapozhnikov (Phys. Faculty, Moscow State University, Leninskie Gory, Moscow 119991, Russia, and Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, olegs2@uw.edu), Dmitry A. Nikolaev, Sergey A. Tsysar (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Vera A. Khokhlova (Phys. Faculty, Moscow State University, Leninskie Gory, Moscow 119991, Russia, and Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

The radiation force balance (RFB) method is a widely accepted approach to characterize ultrasound sources used both in therapy and imaging. Measurement of the radiation force $F$ acting on an absorbing or reflecting target provides the acoustic power $W$. The RFB method is based on the linear relationship $W = Y \cdot c \cdot F$, where $c$ is the speed of sound in the propagation medium and $Y$ is a dimensionless coefficient that depends on the wave structure and reflective properties of the target. The widely accepted expression for a plane wave and an ideal absorber is $Y = 1$. For a spherically focused source the geometric approximation factor $Y = 2/(1 + \cos \alpha)$ is used, where $\alpha$ is the half-aperture angle. The exactness of these and similar approximate relations cannot be checked experimentally because no real sources emit such idealized waves. In the current talk, a more precise approach is presented to determine $Y$ for a given source. To account for the spatial field structure, the acoustic holography method was employed using a 2-D scan with a calibrated hydrophone. In this way, the ultrasound beam's explicitly measured angular spectrum can be used to relate radiation force and acoustic power. [Work supported by NIH R01EB025187 and R01EB007643, and RSF 19-12-00148.]

### Contributed Paper

**10:35**

**2aBAa4. Inverse-filter method to suppress pressure measurement distortion due to spatial averaging across a membrane hydrophone sensitive element.** Keith A. Wear (Ctr. for Devices and Radiological Health, Food and Drug Administration, Bldg 62 Rm. 2114, 10903 New Hampshire Ave., Silver Spring, MD 20993, keith.wear@fda.hhs.gov), Anant Shah, and Christian Baker (National Physical Lab., Teddington, United Kingdom)

This paper reports an investigation of an inverse-filter method to correct for experimental underestimation of pressure due to spatial averaging across a hydrophone sensitive element. The spatial averaging filter (SAF) depends on hydrophone type (membrane, needle, or fiber-optic), hydrophone geometrical sensitive element diameter, transducer driving frequency, and transducer F number (ratio of focal length to diameter) [Wear, *IEEE Trans UFFC* **66**, 318–339 (2019)]. The absolute difference between theoretical and experimental SAFs for 25 transducer / hydrophone pairs was 7% ± 3% (mean ± standard deviation). Empirical formulas based on SAFs allow for correction for hydrophone spatial averaging errors in peak compressional pressure ($p_c$), peak rarefactional pressure ($p_r$), pulse intensity integral (*pii*), mechanical index (MI), and spatial-peak-temporal-average intensity (Istpa). The formulas show, for example, that if a 3-MHz, F/2 transducer is driven to moderate nonlinear distortion and measured at the focal point with a 500-$\mu$m membrane hydrophone, then spatial averaging errors are approximately 16% ($p_c$), 12% ($p_r$ and MI), and 24% (*pii* and Istpa). The formulas are based on circular transducers but also provide plausible upper bounds for spatial averaging errors for transducers with rectangular-transmit apertures, such as linear and phased arrays.

TUESDAY MORNING, 8 DECEMBER 2020 — 9:30 A.M. TO 11:00 A.M.

## Session 2aBAb

## Biomedical Acoustics: General Biomedical Acoustics: Therapeutics & Elastography I

Elisa Konofagou, Chair
*630 West 168th Street, Physicians & Surgeons 19-418, New York, NY 10032*

**Chair's Introduction—9:30**

### Contributed Paper

**9:35**

**2aBAb1. Simulation study of transcranial ultrasound delivery to the hippocampus to test the feasibility of non-invasive neuromodulation.** Xinghao Cheng (Eng. Sci., Univ. of Oxford, Magdalen College, High St., Oxford, Oxfordshire OX1 4AU, United Kingdom, xinghao.cheng@eng.ox.ac.uk), Christopher Butler (Dept. of Brain Sci., Imperial College London, London, United Kingdom), and Robin O. Cleveland (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Transcranial ultrasound stimulation (TUS) is an emerging technology for non-invasive neuromodulation of localised sites in the brain. Here simulations using the k-Wave toolbox, were used to determine an appropriate ultrasound source to target the hippocampus (a relatively deep target in the brain) and the perirhinal cortex which is a neighbouring site that can be used as control for neurostimulation studies. Anonymised T1-weighted MR images were segmented and used as the 3-D simulation space. A virtual source was placed at the target location and captured on a surface outside the head. Phase-conjugation was employed to design a lens for an unfocused transducer and forward propagation employed to predict the acoustic field in the head. The size of the transducer frequency, transducer diameter, and the lens material were each varied. The simulation results suggested that the optimal configuration was a 65 mm diameter, 500 kHz transducer fitted a PDMS lens. Targeting was accurate to within 1 mm with a focal volume less than 30 mm$^3$. In comparison, using a single element transducer with a fixed radius of curvature produce a focal volume greater than 1300 mm$^3$. This motivates developing and validating a lens based TUS with neuromodulation of the hippocampus.

### *Invited Paper*

**9:55**

**2aBAb2. Numerical study of the effect of pressure amplitude and frequency on secondary acoustic radiation force between two polystyrene particles in water.** Azadeh Dashti Cole (MAE, North Carolina State Univ., Eng. Bldg. III (EB3) 4131, Raleigh, NC 27695, adashti@ncsu.edu) and Marie Muller (MAE, North Carolina State Univ., Raleigh, NC)

Poly(N-isopropylacrylamide) microgels emulate the behavior of natural platelets, promoting wound healing. Our previous studies suggested that ultrasound stimulation of these synthetic platelets further enhances wound healing, by increasing particle displacement and deformation. It is necessary to understand and model the interaction between ultrasound and these particles. We hypothesize that the acoustic radiation force may play a role. Two types of radiation force are considered: a primary radiation force caused by the external acoustic field, and a secondary radiation force resulting from scattering between particles. We present a parametric numerical study of the interaction between polystyrene beads (similar to the synthetic platelets) and an acoustic field, to investigate the effect of magnitude, frequency and bandwidth of excitation field on particle response. This is done via the numerical modeling of the primary and secondary radiation forces using 2-D and 3-D FEM. The 3-D simulations capture the re-scattering effects between the particles. One particle is placed at the origin of the coordinate system (scatterer) and a second particle (probe) is placed in the nodal and anti-nodal pressure planes. The numerical results show a good agreement with the governing theoretical equations proposed by Silva and Bruus, suggesting the relevance of the radiation force hypothesis.

### *Contributed Papers*

**10:15**

**2aBAb3. Computational modeling of ultrasound pressures in acoustofluidic channels for enhanced sonoporation.** Connor Centner (Bioengineering, Univ. of Louisville, 580 S Preston St., Louisville, KY 40214, connor.centner@louisville.edu), Chris Holton (Bioengineering, Univ. of Louisville, Louisville, KY), Kavitha Yaddanapudi (Surgery, Univ. of Louisville, Louisville, KY), Thomas Roussel, and Jonathan A. Kopechek (Bioengineering, Univ. of Louisville, Louisville, KY)

Acoustofluidic devices are in development for high-precision cell manipulation and molecular loading applications. In this study, experimental results were compared with computational modeling to characterize the ultrasound pressures within acoustofluidic channels. Acoustofluidic devices were fabricated by integrating PZT transducers into PDMS chips. Viability of Jurkat T cells was measured with MTT assays after acoustofluidic treatment at different ultrasound frequencies between 4.970 and 5.030 MHz. A computational model was implemented using a layered resonator approach to characterize the acoustic pressure profiles within the fluidic channels. The computational results revealed distinct frequencies at which the ultrasound pressures within the channels increased by several orders of magnitude due to constructive interference. Experimental results indicated a significant frequency-dependent difference in cell viability after acoustofluidic treatment (ANOVA p-value = 0.02, n = 10/condition). Cell viability was lowest at 5.000 MHz ($51 \pm 10\%$) compared to other frequencies tested. This trend was consistent with the predicted acoustic pressures within the channels based on the computational model. These results indicate that computational modeling of acoustic pressures within the fluidic channels at specific frequencies can yield important insights for acoustofluidic cell processing applications.

**10:35**

**2aBAb4. CT-based simulation of ultrasound propagation and cavitation in the human body for the assessment of acoustic therapy monitoring in high intensity focused ultrasound.** Kazuki Maeda (Ctr. for Turbulence Res., Stanford Univ., 481 Panama Mall, Stanford, CA 94305, kemaeda@stanford.edu)

*In situ* control of therapy parameters based on real time monitoring of ablation targets is a promising strategy to improve the safety and efficacy of extracorporeal high-intensity focused ultrasound (HIFU). In this work, we present an approach to evaluate the accuracy of acoustic monitoring techniques using computational fluid dynamics simulation. In the numerical setup, computed tomography images are used to model anatomical structures that can scatter and dissipate therapy waves in the body of the patient, including fat, bone, tissues, and organs. An in-house, compressible flow solver is used to simulate ultrasound generation from a transducer, nonlinear focusing of the wave toward therapy targets in the body, and backscattering toward the body surface. The simulation also captures excitation of cavitation bubbles in the focal region and their influence on the pressure fields in the body. Sensing and processing of the scattered signals provide for acoustic monitoring. For demonstration, we apply this approach to HIFU-based lithotripsy that uses burst waves with a focal pressure of $O(1)$ MPa and frequency of $O(100)$ kHz. Accuracies of acoustic monitoring are compared for various sensing and signal processing methods in quantifying the effects of anatomical structures and cavitation on ultrasound irradiation into targeted kidney stones. Finally, we discuss the use of these numerical experiments for control and optimization of therapy parameters. [Work supported by NIH P01-DK043881.]

# Session 2aBAc

## Biomedical Acoustics, Signal Processing in Acoustics, Computational Acoustics, and Physical Acoustics: Modeling and Measuring Nonlinear Ultrasound Signals II

Thomas L. Szabo, Cochair
*Biomedical Engineering, Boston University, 44 Cummington Mall, Boston, MA 02215*

Keith A. Wear, Cochair
*Center for Devices and Radiological Health, Food and Drug Administration, Bldg 62 Room 2114, 10903 New Hampshire Ave., Silver Spring, MD 20993*

**Chair's Introduction—11:15**

### *Contributed Paper*

**11:20**

**2aBAc1. Experimental test of extrapolated acoustic output using single-element transducers.** Timothy A. Stiles (Phys., Kettering Univ., 1700 University Ave., Flint, MI 48504, tstiles@kettering.edu) and Clayton Sparks (Phys., Kettering Univ., Flint, MI)

To ensure the continued safety of diagnostic ultrasound, accurate measurements of the acoustic output from clinical devices are necessary. The effects of nonlinear propagation can cause the current derating method to substantially underestimate relevant values in tissue. One suggested method to overcome this is to measure acoustic output parameters at low levels and extrapolate to values at full output. Tissue mimicking material were produced from concentrated milk. The samples had low scatter but speed of sound, attenuation, and nonlinear parameter B/A consistent with nonfatty soft tissues. Measurements of acoustic output from single element transducers were conducted at a wide variety of transmit levels in both water and through a path consisting mostly of tissue mimicking material. Results of peak rarefactional pressure and MI/MIE using tissue mimicking material at high pressure levels were consistent with extrapolated derated values from low level measurements in water. This agreement was true for the six measured transducers, which spanned a range of center frequencies from 2.0 MHz to 15.0 MHz. These results indicate that it may be possible to use low level transmit pulses in water, derate the measured values, and extrapolate those results to the higher pressure levels typically used in diagnostic ultrasonography.

### *Invited Papers*

**11:40**

**2aBAc2. Measurement and modelling of nonlinear ultrasound fields using Fabry-Perot sensors and k-wave.** Elly Martin (Medical Phys. and Biomedical Eng., Univ. College London, Malet Pl. Eng. Bldg., Gower St., London WC1E 6BT, United Kingdom, elly.martin@ucl.ac.uk), Ben Cox (Dept. of Medical Phys. and Biomedical Eng., Univ. College London, London, United Kingdom), and Bradley Treeby (Univ. College London, London, United Kingdom)

Measurement and modelling of nonlinear ultrasound fields underpins many aspects of clinical ultrasound therapies from transducer design, to treatment planning and computational dosimetry, and quality assurance during the manufacture and use of devices. Measurement of high amplitude nonlinear ultrasound fields presents a significant challenge due to the extreme pressures, temperature rises, and mechanical effects that are generated. Additionally, understanding the properties of suitable sensors and their effect on the measurement is critical to ensure that accurate measurements are obtained. Modelling the propagation of ultrasound under these conditions is similarly challenging, as simulations can become large and computationally expensive when highly nonlinear fields are simulated over clinically relevant domains, and experimental validation becomes more difficult. Here we discuss our approach to overcoming these challenges to perform accurate measurement and modelling of high intensity therapeutic ultrasound fields using the open-source k-Wave toolbox, and devices based on Fabry-Perot interferometers developed for measurement of high acoustic pressures.

12:00

**2aBAc3. System formulation of the nonlinear relaxing wave equation for high-order finite-difference approaches.** Gianmarco Pinton (Biomedical Eng., Univ. of North Carolina at Chapel Hill and North Carolina State Univ., 116 Manning Dr., Mary Ellen Jones Rm. 9212A, Chapel Hill, NC 27599, gia@email.unc.edu)

Finite difference solutions of the wave equation can describe the propagation of waves in heterogeneous media. High-order finite-difference approaches have been used extensively for long-range propagation for seismological applications. For ultrasound imaging, therapy, and neuromodulation, the description of ultrasound propagation must also include attenuation, dispersion, and nonlinearity. Describing these physical phenomena with high-order, stencils has been a persistent challenge. Here we develop a formulation of the nonlinear, attenuating, dispersing, wave equation in system form that is ideally suited for a wide variety of high-order finite-difference stencils. Relaxation functions are modeled in the wave equation by stretching the velocity and pressure variables into a new relaxed coordinate system that describes attenuation, dispersion, and perfectly matched layers using a single formalism. The final system of equations, composed only of first-order derivatives, allows for the implementation of sophisticated stencils. Then, a high-order finite-difference solution of these equations is presented. Solutions of wave propagation are demonstrated for ultrasound imaging applications for deep human abdominal imaging and compared to experimental measurements showing that the numerical method accurately captures the speckle statistics, aberration strength, reverberation strength, and coherence properties. Applications are demonstrated for transcranial neuromodulation, array design, super-resolution, and machine learning.

12:20

**2aBAc4. Factors for validation of measurement-based simulations.** David Sinden (Fraunhofer Inst. for Digital Medicine MEVIS, Fraunhofer MEVIS, Am Fallturm 1, Bremen 28359, Germany, david.sinden@mevis.fraunhofer.de), Srinath Rajagopal, Piero Miloro, and Bajram Zeqiri (National Physical Lab., Teddington, United Kingdom)

In many instances of therapeutic ultrasound it is preferable to produce near-field measurement-based simulations rather than perform direct measurements. For example, simulation from near-field measurements may have lower uncertainties than direct measurements at higher pressures; the entire field can be inspected and differing material properties can be modelled rather than make repeated measurements. However, to properly validate an approach there needs to be an understanding of uncertainties in data acquisition, the limitations of the governing equations and errors from the numerical methods employed. These are not independent, and are primarily determined by a combination of three factors: duration, intensity and inhomogeneity. Criteria for characterizing each factor are presented and consequent procedures outlined. Measurements should consider burst length, positioning from transducer, spatial spacing, required resolution and averaging. From measurements, phase unwrapping and interpolation methods, pseudo-continuous wave approximations, as well as sparse and low-rank methods for data completion or identification of outliers, can be used to characterized the transducer as an initial source condition in an appropriate governing equation. This should consider transmission losses and reflections, frequency-dependent attenuation relations, shock-capturing and -fitting schemes, absorbing boundary conditions and the characterization of source terms for thermal simulations.

# Session 2aBAd

## Biomedical Acoustics: General Biomedical Acoustics: Therapeutics & Elastography II

Elisa Konofagou, Chair
*630 West 168th Street, Physicians & Surgeons 19-418, New York, NY 10032*

**Chair's Introduction—11:15**

### *Contributed Papers*

**11:20**

**2aBAd1. Analytical solution based on spatial distortion for a time-harmonic dipole source in a transverse isotropic viscoelastic solid.** Thomas Royston (Univ. of Illinois at Chicago, 851 S. Morgan St., Rm. 218 SEO (MC 063), Chicago, IL 60607, troyston@uic.edu)

Elastography refers to mapping mechanical properties in a material by measuring wave motion in it using noninvasive optical, acoustic or magnetic resonance imaging methods. Elasticity and viscosity can depend on location and direction. A material with aligned fibers may have different elastic and viscosity values along the fibers versus across them. Reconstruction refers to converting wave measurements into a mechanical property map. To make reconstruction analytically tractable, isotropy and homogeneity are often assumed. But, isotropic homogeneity is often not the case of interest, when there are pathological conditions or hidden anomalies non-uniformly distributed in fibrous or layered structures. A strategy of spatial distortion to make an anisotropic problem become isotropic has been previously validated in two-dimensional transverse isotropic (TI) viscoelastic cases. Here, the approach is extended to the three-dimensional problem by considering a time-harmonic point force (dipole) in a TI viscoelastic material. The resulting wave field, exactly solvable using a Radon transform with numerical integration, is approximated via spatial distortion of the closed form analytical solution to the isotropic case. Different distortions are used depending on whether or not the polarization of the wave motion is orthogonal to the axis of isotropy resulting in differing levels of accuracy.

**11:40**

**2aBAd2. Simulating insertion-based measurement of phantom attenuation and membrane transmission loss.** Karthik J. Nagabhushana (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, 306 North Wright St., Urbana, IL 61801, kjn3@illinois.edu), William D. O'Brien, and Aiguo Han (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, Urbana, IL)

Attenuation of reference phantoms that are used in quantitative ultrasound applications must be accurately measured. In this regard, we recently proposed an insertion-based planar reflection technique to simultaneously estimate membrane transmission loss and phantom attenuation. Simulating the measurement procedure can assist with validating and improving the methodology and gauge its accuracy and precision by carefully controlling the simulation's setup parameters. For simulation, a pulse echo setup with a single element transducer immersed in water was implemented in k-Wave. First, a reference plane with known acoustic properties is positioned at the focus, yielding the reference echo that models transducer diffraction and frequency response. Next, a phantom covered by a thin layer of acoustic windowing membrane is inserted between the transducer and reference plane, yielding the insertion echo that captures the phantom attenuation and membrane transmission loss. Finally, the phantom top surface, lined with the membrane, is positioned at the focus, thereby yielding the surface echo that captures the membrane reflectivity. Spectral analysis of the three echoes estimates the membrane transmission-loss-corrected phantom attenuation. Preliminary simulation of a phantom with predefined attenuation coefficient of 0.6–0.9 dB/cm MHz yielded a root mean square error of 0.06 dB/cm MHz over 3–5 MHz. [NIH Support: R01CA226528, R01DK106419, and R01HD089935.]

**12:00**

**2aBAd3. A soft-computational method for imaging nonlinear mechanical properties of tissue-like media using ultrasound.** Yiliang Wang (Dept. of Mech. Sci. and Eng., Univ. of Illinois at Urbana-Champaign, 405 N. Mathews M/C 251, Urbana, IL 61801-2325, wang513@illinois.edu), Cameron Hoerig (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res. Inst., New York, NY), Jamshid Ghaboussi (Dept. of Civil and Environ. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), and Michael Insana (Dept. of Bioengineering, Univ. of Illinois at Urbana-Champaign, Urbana, IL)

The Autoprogressive (AutoP) Method is a data-driven technique for estimating stress-strain maps from force-displacement measurements. It separately inputs the quasi-static compression force applied to tissues by an ultrasound transducer and the corresponding displacements estimated from a speckle tracking routine into two finite-element algorithms (FEAs) connected through one mesh. The FEAs are associated to each other through neural network constitutive models (NNCMs) that iteratively learn a non-parametric mapping from strain to stress. After training, the stress-strain relationship learned by a NNCM can be compared to known constitutive models to estimate material properties of soft tissues. Here, we apply Veronda-Westmann constitutive model (VWCM) to the fully trained AutoP results to yield images of shear moduli and the nonlinear parameter. We tested AutoP on data acquired from homogeneous, nonlinear, tissue-like media under conditions for which the VWCM applied. Comparing data-driven AutoP results with model-based estimates, we validated the approach. AutoP was then applied to heterogeneous phantoms with inclusions made of linear-elastic materials embedded in nonlinear-elastic backgrounds, where mechanical properties of both media were independently verified. Images of the shear modulus and nonlinear parameter under a wide range of quasi-static loads show that AutoP has great potential for imaging nonlinear tissues *in vivo*.

12:20

**2aBAd4. The generalized finite amplitude insert substitution method for estimation of the ultrasound parameter of nonlinearity.** Anastasiia Panfilova (Elec. Eng., Eindhoven Univ. of Technol., Groene Loper 19, Eindhoven, North Brabant 5612 AR, The Netherlands, A.P.Panfilova@tue.nl), Xufei Chen, Ruud J. Sloun, Hessel Wijkstra (Elec. Eng., Eindhoven Univ. of Technol., Eindhoven, The Netherlands), Oleg A. Sapozhnikov (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation and Appl. Phys. Lab., Univ. of Washington, Moscow, Russian Federation), and Massimo Mischi (Elec. Eng., Eindhoven Univ. of Technol., Eindhoven, The Netherlands)

The estimation of the ultrasound parameter of nonlinearity (B/A) has shown promise for improved tissue characterization. The finite amplitude insert substitution method (FAIS) is a simple technique that measures B/A. It derives B/A based on the ratio between the 2nd harmonic generated in a reference medium (known B/A) and in the sample of interest, measured in transmission mode between a source and a receiver transducer. The classical FAIS was presented for media with a linear frequency dependence of attenuation, and assumes the sample to be positioned close to the receiver. In this work, the FAIS was generalized to allow arbitrary sample positions and arbitrary frequency dependence of attenuation. The introduced generalization of the method and the utilized setup for B/A measurement were validated by measuring B/A in corn oil and 2 animal tissues: porcine liver and porcine fat, whose B/A are known. The experimental conditions which optimize the accuracy of the estimated B/A were also investigated. The mean B/A for the chosen measurements was $10.8 \pm 0.8$ for oil (ref. B/A = 10.3), $5.1 \pm 0.8$ for porcine liver (ref. B/A = 6.9) and $11.9 \pm 1.6$ for porcine fat (ref. B/A = 10.9), showing the feasibility of B/A measurement by the proposed generalized method.



TUESDAY MORNING, 8 DECEMBER 2020 — 9:30 A.M. TO 11:00 A.M.

## Session 2aCAa

## Computational Acoustics, Physical Acoustics, Structural Acoustics and Vibration, and Underwater Acoustics: Ray Methods Across Acoustics I

Michelle E. Swearingen, Cochair
*Construction Engineering Research Laboratory, U.S. Army ERDC, PO Box 9005, Champaign, IL 61826*

Jennifer Cooper, Cochair
*Applied Physics Laboratory, Johns Hopkins University, 11100 Johns Hopkins Rd., Mailstop 8-220, Laurel, MD 20723*

**Chair's Introduction—9:30**

### *Invited Papers*

9:35

**2aCAa1. Ray methods and infrasound.** David Norris (National Security Solutions, ENSCO, Inc., 4849 N. Wickham Rd., Melbourne, FL 32940, norris.david@ensco.com)

Infrasound is formally defined as sound at frequencies of 20 Hz and below. Because of low atmospheric absorption, infrasound can propagate thousands of km or more with minimal attenuation. Ray methods are effective in characterizing long-range infrasound propagation, in particular for predicting acoustic paths and identifying dominant ducting mechanisms. In this talk, the assumptions, advantages, and disadvantages of infrasound ray modeling will be reviewed. Focus will be given to the effects of moving media, range-dependence, and horizontal refraction. Examples from special event studies will be shown to illustrate specific effects. The talk will conclude with the author's view of the modeling challenges facing the current infrasound research community moving forward.

**9:55**

**2aCAa2. Advanced sonic boom propagation model based on a single-ray Jacobian.** Joel B. Lonzaga (Structural Acoust. Branch, National Aeronautics and Space Administration, 2 N. Dryden St. (MS 463), Hampton, VA 23681, joel.b.lonzaga@nasa.gov)

A sonic boom is a nonlinear event whose propagation can be modeled involving two stages. The first stage consists of ray path calculations using linear approximations, while the second stage deals with a nonlinear transport of the acoustic energy along these paths. This presentation discusses a second-order finite difference numerical approach used to predict the ray paths in range-dependent atmospheres and compares it to existing finite difference schemes. The approach is validated using exact solutions obtained for stratified atmospheres. In the second stage, the geometrical spreading effect needed in determining sonic boom waveforms from a Burgers' equation is obtained using a method that only needs a single ray rather than four rays required by most existing sonic boom propagation codes. The single-ray method calculates the Jacobian, associated with the coordinate transformation from a suitable ray coordinate system to the Cartesian coordinate system, directly from the ray tracing equations. While the four-ray method approximates the geometrical spreading using a finite difference scheme involving the four rays, the single-ray method does not rely on this approximation and instead depends purely on the acoustical kinematic properties of the atmosphere. Comparisons of results using these two methods are discussed.

## *Contributed Papers*

**10:15**

**2aCAa3. Ray tracing for efficient simulation of curved sound propagation paths: Towards real-time auralization of aircraft noise.** Philipp Schäfer (Inst. of Tech. Acoust., RWTH Aachen, Kopernikusstraße 5, Aachen 52074, Germany, philipp.schaefer@akustik.rwth-aachen.de) and Michael Vorlaender (Inst. of Tech. Acoust., RWTH Aachen, Aachen, Germany)

Thanks to modern auralization methods, simulated aircraft flyovers can be experienced in virtual environments. Listening to such an auralization provides a more sophisticated impression of annoyance of aircraft noise in real scenarios than reading averaged sound pressure levels or noise maps. For proper auralization results, translation and refraction due to wind and to inhomogeneity of the atmosphere should be taken into account, which leads to curved sound propagation paths. However, state of the art auralization tools including these effects do not allow real-time simulations on CPUs. In this work, an efficient ray tracing tool for simulating curved propagation paths between an aircraft and a listener is introduced. The performance and limitations are discussed for a typical aircraft trajectory. The benchmark results suggest that real-time auralization of aircraft noise using ray tracing is possible if the processing of propagation paths in the overlying auralization framework is extended accordingly.

**10:35**

**2aCAa4. Simulations of X-59 low-booms propagated through measured atmospheric profiles in Galveston, Texas.** William Doebler (NASA Langley Res. Ctr., NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, william.j.doebler@nasa.gov)

During NASA's Quiet Supersonic Flights 2018 (QSF18) test, residents of Galveston, Texas and surrounding communities were exposed to low amplitude sonic booms for a community noise survey. Despite the flight test lasting only 11 days, substantial weather variability occurred including snow and record high temperatures. Similar atmospheric variability could occur in future community noise flight tests with the X-59 Quiet Supersonic Technology aircraft that is currently being constructed. X-59 is designed to have a Stevens Mark VII Perceived Level (PL) of 75 dB in a standard atmosphere. In this work, X-59's low-boom performance in real atmospheres was tested by using NASA's PCBoom code to simulate propagation of its nearfield pressure through the atmospheric profiles measured during QSF18. Results indicate undertrack booms would not exceed 75 PLdB in these atmospheres. A range of about 8.5 PLdB between the loudest and quietest undertrack boom was observed indicating adjustments to the X-59 flight condition may be needed to achieve target loudness levels during future sonic boom community noise surveys. Attenuation rate, ray tube area, path length and other quantities throughout propagation of the loudest and quietest booms are presented, which indicate humidity differences below 15kft were a primary driver of the PL differences.

## Session 2aCAb

# Computational Acoustics, Physical Acoustics, Structural Acoustics and Vibration, and Underwater Acoustics: Ray Methods Across Acoustics II

Michelle E. Swearingen, Cochair
*Construction Engineering Research Laboratory, U.S. Army ERDC, PO Box 9005, Champaign, IL 61826*

Jennifer Cooper, Cochair
*Johns Hopkins University Applied Physics Laboratory, 11100 Johns Hopkins Rd., Mailstop 8-220, Laurel, MD 20723*

**Chair's Introduction—11:15**

### *Invited Papers*



**11:20**

**2aCAb1. Ray methods in architectural acoustic design.** Nicolaus T. Dulworth (Threshold Acoust., 141 W Jackson Blvd, Ste. 2080, Chicago, IL 60604, ndulworth@thresholdacoustics.com), John T. Strong, and Carl Giegold (Threshold Acoust., Chicago, IL)

Raytracing is a simple but powerful tool in acoustics consulting because of its ability to geometrically approximate wave behavior across a large portion of the audible spectrum. The use of this method requires an understanding of the frequency-dependent interaction between sound waves and geometry, which is a wave-specific behavior that any use of ray-based approximations must consider. The limitations of these approximations may necessitate geometric modifications in a model to render credible results, or may even dictate raytracing is inappropriate for a given application. Ray-based analysis can be used for applications with a wide range of complexity, from hand-traced sketches to computational simulation for full-scale rooms. Common applications include analysis of reflector coverage, design of room shape and major room components, and impulse response simulation for analysis and auralization. Due to their efficiency and simplicity, ray methods remain indispensable in architectural acoustic analysis.

**11:40**

**2aCAb2. Ray model limitations.** Ruth E. Keenan (SISL, Appl. Res. Labs., 10000 Burnet Rd., Austin, TX 78758-4423, rkeenan@arlut.utexas.edu)

Ray based models have been used extensively in underwater acoustics with the generalization that they are only valid at high frequency. This presentation will demonstrate that frequency is not the critical parameterization for the Gaussian beam ray models, Bellhop and Comprehensive Acoustic Simulation System/Gaussian Ray Bundle (CASS/GRAB). A more useful parameter to determine the expected accuracy are the number of modes trapped in the waveguide of interest. These Gaussian beam ray methods tend to break down when the waveguide contains fewer than 5 modes. Surface duct environments can be particularly challenging, especially when there is a significant amount of energy that diffracts or leaks out of the duct. Lower frequencies will reach the low order mode limit as water depth decreases. Yet, these ray methods can be accurate for long distance, low frequency waterborne propagation and can provide important angular and temporal resolution of arrival structures.

**12:00**

**2aCAb3. Ray and beam tracing in underwater acoustics.** Michael B. Porter (Heat, Light, and Sound Res., Inc., 12625 High Bluff Dr., STE 211, San Diego, CA 92130-2054, Porter@HLSResearch.com)

In 1919, Lichte published a paper on the influence of horizontal temperatures layers in sea water (his words). This paper is probably the first to use rays to understand sound propagation in the ocean and interestingly also predicts that deep water will provide "significantly greater ranges than in shallow water" anticipating later important discoveries about the deep-water sound channel. It is striking that ray modeling could be done in 1919. In the 1960s when digital computers assumed their central role in sound propagation, ray models were the inspiration. Over the following years many ray models were developed and every significant laboratory doing underwater acoustics had developed one. However, ray modeling is simple on one level, yet incredibly complex on others. A Navy report from 1985 concluded that they all had significant problems. In the following 30-odd years, other modeling approaches were pursued more aggressively; however, ray models had unique advantages and continued to evolve into modern beam-tracing methods. These new models are not free from the artifacts inherent in the ray approximation, but nevertheless are very powerful and significantly more accurate and reliable than their ancestors. Today they work. This talk will discuss the history and the state of the art.

*Contributed Paper*

**12:20**

**2aCAb4. Comparison of ray tracing to energy flux for computing transmission loss in uncertain environments.** Sheri Martinelli (Appl. Res. Lab., The Penn State Univ., PO Box 30, M/S 3230D, State College, PA 16804, slm77@psu.edu)

The Lagrangian nature of the ray tracing equations imparts a known sensitivity of solutions to initial conditions for long integration times. Yet ray models remain a standard for high frequency wave propagation due to their computational efficiency and amenability to parallelization. Acoustic energy flux methods can provide an alternative solution in the high frequency regime when a large number of propagating modes are closely spaced in angle. The energy flux approach has also been shown to be quite robust to variability in bathymetry and sediment type in underwater applications. In this work, we apply generalized polynomial chaos expansions to represent both an acoustic ray model and an acoustic energy flux model in identical underwater environments in which the sound speed profile, source depth, and bathymetry are uncertain and thus modeled by random quantities. The output quantity of interest is taken to be the coherent (semi-coherent in the case of energy flux) transmission loss as a function of range and depth. Comparisons are made through assessment of the statistical moments of the output, and a sensitivity study on the uncertain parameters.

TUESDAY MORNING, 8 DECEMBER 2020 — 11:15 A.M. TO 12:10 P.M.

## Session 2aED

## Education in Acoustics: Acoustics Education Prize Lecture

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

**Chair's Introduction—11:15**

*Invited Paper*

**11:20**

**2aED1. Over a decade of decibels—Celebrating teaching architectural acoustics within an architecture curriculum to students with various majors and minors.** Daniel Butko (Architecture, The Univ. of Oklahoma, 830 Van Vleet Oval, Gould Hall 265, Norman, OK 73019, butko@ou.edu)

Having grown up with a musically inclined father who inspired an early appreciation for acoustically related environments and studying under Gary Siebein at UF, the author bridges architecture and acoustics in both teaching and practice. Adjunct positions focused on environmental technologies and acoustic-based courses at UF and K-State led to the current position of Associate Professor of Architecture and OU Gibbs College of Architecture Research Fellow at The University of Oklahoma. Since beginning at OU in 2010, course assignments such as dedicated architectural acoustics electives, design studios, materials and environmental systems, and independent studies have integrated acoustics into curriculum objectives. With over 1300 students to date, majors include Architecture, Architectural Engineering, Environmental Design, Interior Design, and Pre-med with minors in Anthropology, Chemistry, Computer Science, Construction Management, Literature, Business Management, Geography, Foreign Languages, History, Mathematics, and Sociology. Opportunities for pedagogical applications include: hands on experiments; field measurements and data collection; study of musical instruments; precedent studies; laboratory and manufacturing facility tours; and material prototype fabrication with industry partners and other colleges. This presentation showcases how teaching broad spectrum acoustical topics resulted in acoustical integration within capstone projects, student coauthored publications and presentations, and student recognition.

## Session 2aMU

# Musical Acoustics and Education in Acoustics: Musical Acoustics Education at the Undergraduate Level I

Andrew C. Morrison, Chair
*Natural Science, Joliet Junior College, 1215 Houbolt Dr., Joliet, IL 60431*

**Chair's Introduction—11:15**

## *Invited Papers*

**11:20**

**2aMU1. Strategies for teaching a musical acoustics class online.** Andrew A. Piacsek (Dept. of Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926-7422, andy.piacsek@cwu.edu)

Physics of musical sound is a first-year undergraduate course at Central Washington University that has been offered for over 20 years. The course is normally taught in a studio classroom format that utilizes peer instruction. Over the past few years, much of the course content and many of the exercises have been adapted to an online course management system (Canvas), enabling a significant degree of flipped teaching. Instead of a textbook, extensive use is made of online resources, including simulations, sounds, and free audio software. In 2018, I took this a step further and developed a fully online version of the course to be offered during the summer session. The goal was to replicate many elements of the traditional in-person class, including student interactions and facilitating a final project that involves constructing or analyzing a musical instrument. This presentation will emphasize the design and efficacy of three particular class elements: (1) online discussions based on a reading or video; (2) small group projects, including lab activities; (3) a term project culminating in the design and construction of a musical instrument.

**11:40**

**2aMU2. Playing to learn: Hands-on activities in the musical acoustics classroom.** Laurie McNeil (Phys. & Astronomy, Univ. of North Carolina at Chapel Hill, Phillips Hall, CB #3255, Chapel Hill, NC 27599-3255, mcneil@physics.unc.edu)

The physics of musical instruments is a topic that often appeals to students who might not otherwise take a physics course. These students (and the physics majors!) respond very well to active-learning approaches in which they are led to explore the mathematics and physics they are learning by means of real objects making real sounds. I will describe the activities I have incorporated into a course for first-year undergraduates that I have co-taught with a colleague from the Music Department for two decades. Students demonstrate phenomena on instruments they play, make measurements of vibrating strings and air columns, produce vibrating reeds from drinking straws, design wind chimes, and build novel (and often comical) stringed and wind instruments out of found objects. The class culminates in a concert in which students perform their own compositions for ensembles of those instruments. I will relate the success of these activities to the findings from physics education research that underpin them.

**12:00**

**2aMU3. Techniques that can be used in a musical acoustics course.** Gordon P. Ramsey (Phys. Dept., Loyola Univ. Chicago, Chicago, IL 60660, gramsey@luc.edu)

A course in Musical Acoustics provides a unique opportunity to involve some innovative techniques in and out of the classroom. Our Physics of Music course is a thorough integration of physics and music.[1] It starts with the mathematical structure of music and discusses musical styles and how they differ. After an introduction of related physics concepts, the physical structure and operation of instruments in the various groups are studied. Connection is made of the instruments and how they reproduce the mathematical nature of music. The course integrates different learning modes, some of which are not found in a typical physics course. The classes include lecture/demonstration, group discussions, in-class laboratories, guest lecturers and a final individual project encompassing many course elements. The constant connection between the physics and the music, along with varied learning techniques, including hands-on experience, provides a motivating approach for students to experience science in a familiar context. This talk will provide a brief course overview and discussion of the non-standard instruction elements used in the course. [1] For a detailed course description, see *POMA*18 (2014). DOI: 10.1121/1.4895817

12:20

**2aMU4. Teaching musical acoustics with simple models and active learning.** Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., N251 ESC, Provo, UT 84602, tbn@byu.edu), Cameron T. Vongsawad (Phys. & Astronomy, Brigham Young Univ., Orem, UT), and Stephen G. Onwubiko (none, Enugu, Nigeria)

When teaching musical acoustics to undergraduates, some have difficulty understanding how acoustical principles are related to music production. These difficulties are mitigated by using simple physical models and employing active learning activities. Three models are particularly helpful: mass-spring systems, string and tube resonances, and Helmholtz resonators. Students can gain hands-on experience with these models in both personal and guided laboratory experimentation. Examples of both types of experiments are provided to illustrate how they enable students to better understand the acoustics of musical instruments. These active learning activities highlight similarities and differences between the different families of musical instruments. The active learning activities are particularly impactful when students write about their experiences using proper acoustical terminology. The resulting understanding of acoustical principles enhances music education.

TUESDAY MORNING, 8 DECEMBER 2020 — 9:30 A.M. TO 10:40 A.M.

## Session 2aNSa

## Noise: Forty-One Years of Responding to External Stimuli: A Session in Honor of Elliott Berger I

Cameron J. Fackler, Cochair
*3M Personal Safety Division, 7911 Zionsville Rd., Indianapolis, IN 46268*

Lauraine L. Wells, Cochair
*3M, 817 W. 4th St., Loveland, CO 80537*

William J. Murphy, Cochair
*Division of Field Studies and Engineering, Noise and Bioacoustics Team, National Institute for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998*

**Chair's Introduction—9:30**

### *Invited Papers*

9:35

**2aNSa1. Forty-one years of responding to external stimuli: Elliott Berger's career in hearing conservation.** Cameron J. Fackler (3M Personal Safety Div., 7911 Zionsville Rd., Indianapolis, IN 46268, cameron.fackler@mmm.com) and Jeffrey L. Hamer (3M Personal Safety Div., Indianapolis, IN)

It was 1976, hearing conservation and earplug technology had just been advanced by the introduction of the first slow-recovery roll-down foam earplug four years ago, and Elliott Berger had just entered the workforce. Over the next 41 years, Elliott would be a key player in many other advancements in hearing conservation. This paper will briefly summarize his career and contributions to hearing conservation, including research and development of new hearing protectors, standardization of many aspects of hearing conservation, and education of hearing conservation practitioners. Along the way, Elliott documented his work in numerous journal articles, book chapters, and presentations. Several of Elliott's colleagues, collaborators, and friends are here today to fill in more of the details of his career and work to advance hearing conservation.

**9:55**

**2aNSa2. The influence of Elliott Berger in shaping occupational hearing loss prevention.** Dennis Driscoll (Driscoll Acoust., 2560 S. Orchard St., Lakewood, CO 80228, dennis@driscollacoustics.com)

It was at the June 1979 ASA meeting in Cambridge where I first me Elliott Berger. We were introduced by Larry Royster, who was our graduate professor at N.C. State University. Elliott had long departed before I started my graduate program, but the echoes of his past still reverberated throughout Larry's acoustics lab. I will elaborate more during my talk. I started my professional career in occupational noise in mid-1980. For almost the next forty years I worked with Elliott on several professional association committees, participated in countless seminars and conferences, toiled over the *AIHA Noise Manual*, 5th Edition (and pending $6^{th}$), and have enjoyed his friendship since that day in 1979. As a scientist, Elliott has consistently and steadfastly worked to assure and improve the protection of individuals from occupational noise. As a peer, he has provided mentorship to not only me, but many of the experts in this field. The topic of my talk will cover a summary of Elliott's prolific contributions to the cause of hearing loss prevention.

**10:15**

**2aNSa3. The Noise MANual.** Deanna K. Meinke (Audiol. & Speech-Lang. Sci., Univ. of Northern Colorado, 501 20th St., Greeley, CO 80639, Deanna.Meinke@unco.edu) and Richard L. Neitzel (Environ. Health Sci., Univ. of Michigan, Ann Arbor, MI)

The name Elliott Berger has appeared on the covers of the 4th, 5th, and 6th editions of the American Industrial Hygiene Association (AIHA) *Noise Manual* since 1986. As principle editor for the 4th and 5th editions and co-editor of the 6th edition (2020), Elliott's contributions extend beyond the written words, but to the personal connections that attracted premier authors to contribute chapters. The 4th edition was one of AIHA's best selling texts with over 13 000 copies in print, and the 800 page 5th edition (2000) was selected by the AIHA Publications Committee as "Outstanding 2000 Technical Committee Publications," and as "Top-Selling Technical Committee Publication for the years 2000 and 2001. The chapters have reached readers from multiple disciplines beyond the AIHA industrial hygiene membership, including occupational health and safety, medicine, nursing, audiology, engineering and public health. The chapters are used in college courses and research labs globally to train future professionals in hearing conservation. This accomplishment is noteworthy in terms of the dedication, expertise, precision and perseverance that Elliott exhibited over the past 34 + years. We're honored to give tribute to the "Noise MANual." Conflict of Interest Statement: The presenters are co-editors of the 6th Edition of the Noise Manual.

## Session 2aNSb

# Noise: Forty-One Years of Responding to External Stimuli: A Session in Honor of Elliott Berger II

Cameron J. Fackler, Cochair
*3M Personal Safety Division, 7911 Zionsville Rd., Indianapolis, IN 46268*

Lauraine L. Wells, Cochair
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William J. Murphy, Cochair
*Division of Field Studies and Engineering, Noise and Bioacoustics Team, National Institute for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998*

**Chair's Introduction—11:15**

## *Invited Papers*

**11:20**

**2aNSb1. Elliott Berger: A good scientist to have around!.** mead c. killion (Etymotic Res., 61 Martin Ln., Elk Grove Vlg, IL 60007, abonso@aol.com)

The teachings of Elliott Berger are always clear. His recent paper "The Art of Presentation" explains his approved method. He has been the direct and indirect mentor to most of us interested in protecting hearing, and he is *the* expert in the real-ear performance of hearing protection devices. His 21 EARLogs® provide a treasure trove of information. The Berger *et al. Noise Manual* has been without peer in several editions. His integrity is absolute, his enthusiasm knows few bounds, and his determination to get to the bottom of vexing questions has produced some of the most important information in our field. A colleague and I once visited Elliott hoping to sell him on a new ready-to-wear earplug with a flat 20 dB attenuation. He and Bob Falco liked the performance, but opined that it was *very ugly*—and suggested a more attractive construction which produced the *same performance*. The resulting ER20 HiFi earplug has sold some three million pairs between the two companies: Hundreds of thousands of drummers and band students have normal hearing today as a result. Elliott sometimes has fun even with an important question: "So, how do you want your NRRs: Realistic or sunny-side-up?"

**11:40**

**2aNSb2. Elliott Berger's role in the standardizing measurements of hearing protection devices.** William J. Murphy (Div. of Field Studies and Eng., Noise and Bioacoustics Team, National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov) and Lauraine L. Wells (3M Co. Personal Safety Div., 3M Co., St. Paul, MN)

Elliott Berger has been perhaps the most influential person in the development of standards for hearing protection devices within the Acoustical Society of America and the broader international standards community. He became involved with the American National Standards Institute (ANSI) S3 working group 10 shortly after he joined Cabot Safety Corporation in 1976. In 1985, Elliott assumed the chair position of ANSI S12 working group 11, which was responsible for the ANSI S12.6, *American National Standard Methods for Measuring the Real-ear Attenuation of Hearing Protectors*. He worked on developing a rating method that could be consistently implemented in testing labs and which represented the performance of hearing protectors. Thirty-five years later, the methods are largely the same, but have seen a harmonization between ANSI and international standards. This paper will discuss the development of these standards, methods to measure passive, level-dependent, and active noise cancellation hearing protection devices, and methods to qualify hearing protection fit-test systems. Elliott's dedication, meticulous commitment to quality, and leadership have forged consensus amidst diverse interests, leaving a lasting impact on occupational hearing loss prevention.

12:00

**2aNSb3. Elliott Berger's contributions to military hearing conservation and standards.** Richard L. McKinley (none, 4366 Osborn Rd., Medway, OH 45341, rich3audio@aol.com)

Military organizations world-wide experience the great majority of the extreme occupational noise exposures. The noise sources range from tanks and armored personnel carriers to fighter jets and shoulder launched missiles. It is a difficult task to protect the men and women who serve their countries and experience these extreme environments where the continuous noise levels can reach 150 dBA and the peak noise levels from firing weapons can reach 195 $dB_{peak}$. Stable methods to measure hearing protector attenuation benefit the development and optimization of hearing protection. National and international standards measuring hearing protection attenuation have been the focus of much of Elliott's work during his career and now in semi-retirement. These standards have been the catalysts that have enabled the development of improved hearing protectors that help conserve the hearing of service members in military noise environments. Data will be presented showing ranges of military continuous and impulsive noise exposures, the amount of hearing protection attenuation needed to meet US DoD noise exposure criteria, and the ANSI standards for the measurement of hearing protector attenuation.

## *Contributed Paper*

12:20

**2aNSb4. Comparison of two types of earplugs by musicians.** Vishakha Rawool (Commun. Sci. & Disord., Univ. of MS, 1006 Briarwood Dr., Oxford, MS 38655, vishakharawool8@gmail.com) and Roraine Bunag (The Centers for Adv. ENT Care, Bowie, MD)

This research compared the preference of musicians across two earplugs; a typical earplug with un-even attenuation across the frequency range (Clarity 695) and another with relatively flat attenuation (ER-20). Thirty-three musicians with normal hearing were first oriented to hearing loss, benefits of using earplugs and the correct procedures for inserting earplugs. Then, they played music for three minutes in five conditions (Trial 1 with First earplug type, Trial 1 with second earplug type, no earplugs, Trial 2 with first earplug type, Trial 2 with second earplug type) and completed a Visual Analogue Scale (VAS). The VAS scores for the ER-20 earplugs were significantly higher and a significantly higher percentage of musicians preferred the ER-20 earplugs (74%) over the Clarity earplugs (26%). This preference may be related to the relatively flat attenuation provided by the ER-20 earplugs across the frequency range or the lower Noise Reduction Rating (NRR:13) of this earplug compared to the clarity earplugs (NRR:21). The VAS scores improved on trial 2 over trial 1 suggesting some adaptation to the use of earplugs over time. [Funded by the 2015 – 2016 Wirt C. and Mae S. Belcher Graduate Education Award at West Virginia University.]

## Session 2aPAa

# Physical Acoustics, Musical Acoustics, and Biomedical Acoustics: Acoustical Measurements Through Optical Principles I

Gregory W. Lyons, Cochair
*Information Technology Laboratory, U.S. Army Engineer Research and Development Center, 3909 Halls Ferry Road, Vicksburg, MS 39180*

Thomas R. Moore, Cochair
*Department of Physics, Rollins College, Box 2743, Rollins College, Winter Park, FL 32789*

**Chair's Introduction—9:30**

## *Invited Papers*

**9:35**

**2aPAa1. The history of holographic vibration analysis.** Karl A. Stetson (Karl Stetson Assoc. LLc, 2060 South St., Coventry, CT 06238, kastetson@holofringe.com)

This paper describes the evolution of holographic vibration analysis from its discovery in 1964 to its present form as a technique of digital holography. The speckle interferometer is discussed as a replacement for real-time holographic interferograms and electronic speckle pattern interferometry (ESPI) as the precursor to digital holography. The development of methods for converting vibratory (Jo) fringe patterns into numerical data is discussed along with applications to Modal Assurance Criteria (MAC) calculations. Finally, an application is presented for vibratory strain measurement in one dimension.

**9:55**

**2aPAa2. Constructing an electronic speckle pattern interferometry system for visualizing operational deflection shapes.** Andrew C. Morrison (Natural Sci., Joliet Junior College, 1215 Houbolt Dr., Joliet, IL 60431, amorriso@jjc.edu) and Thomas R. Moore (Dept. of Phys., Rollins College, Winter Park, FL)

Electronic speckle pattern interferometry (ESPI) is an effective optical method for visualizing operational deflection shapes of vibrating objects. The time-averaged ESPI method is used to show how an object vibrates when driven sinusoidally. We describe the construction of our system, which we built using standard optical components. Such a system is appropriate for both research and educational purposes. We will discuss the lessons learned by constructing several ESPI systems.

**10:15**

**2aPAa3. Variations on the common electronic speckle pattern interferometer.** Thomas R. Moore (Dept. of Phys., Rollins College, Box 2743, Rollins College, Winter Park, FL 32789, tmoore@rollins.edu)

Electronic speckle pattern interferometry is often used to visualize deflection shapes of vibrating objects. We discuss variations on this common laboratory technique, emphasizing unique applications as well as methods for reducing the cost and complexity of the experimental arrangement.

**10:35**

**2aPAa4. Acoustic excitation of landmines and remote sensing from a moving truck.** Brad Libbey (CCDC C5ISR Ctr. NVESD, U.S. Army, 10221 Burbeck Rd., Ft. Belvoir, VA 22060, brad.libbey2.civ@mail.mil)

The U.S. Department of Defense developed a 960 pixel laser Doppler vibrometer to capture seismic vibration and detect buried landmines. The sensor relied on a loudspeaker to create the seismic disturbances. This system captured seismic spatial responses while advancing at 1 m/s at 30 m standoff. Automated target recognition algorithms quickly processed the spatial vibration data to warn operators of threats. Sensor noise was highly variable, but on the order of .05mm/s in a band 50–400 Hz over which the sensor operated. Inertial sensing reduced gross Doppler components common to all channels. Even with these corrections, channel dropouts remained a challenge, so filtering of erroneous samples in time and space was required to improve the data quality, which is possible due to 960 simultaneous spatial samples. The large pixel count permitted road scans of 2 m $\times$ 1 km in less than 30 min. The system proved effective at finding buried threats that produced seismic anomalies at the surface, but additional challenges lie in differentiating target responses from clutter. This presentation will introduce the system and display high-resolution seismic images excited acoustically. It will also illustrate an acoustic wave captured directly by the lasers as the air's index-of-refraction fluctuated.

# Session 2aPAb

## Physical Acoustics, Musical Acoustics, and Biomedical Acoustics: Acoustical Measurements Through Optical Principles II

Gregory W. Lyons, Cochair
*Information Technology Laboratory, U.S. Army Engineer Research and Development Center, 3909 Halls Ferry Road, Vicksburg, MS 39180*

Thomas R. Moore, Cochair
*Department of Physics, Rollins College, Box 2743, Rollins College, Winter Park, FL 32789*

**Chair's Introduction—11:15**



### *Invited Papers*

**11:20**

**2aPAb1. Laser Doppler multi-beam differential vibrometry.** Vyacheslav Aranchuk (National Ctr. for Physical Acoust., Univ. of MS, 145 Hill Dr., University, MS 38655, aranchuk@olemiss.edu), Ina Aranchuk, Brian Carpenter, and Craig J. Hickey (National Ctr. for Physical Acoust., Univ. of MS, University, MS)

A Laser Multi-Beam Differential Interferometric Sensor (LAMBDIS) for measuring vibration fields has been developed that alleviates one of the major issues of traditional laser Doppler vibrometers: effect on measurements by motion of the vibrometer itself. The LAMBDIS simultaneously measures Doppler shifts of light reflected from different points on the object surface illuminated with a linear array of laser beams. As a result, the LAMBDIS measures relative velocities between points on an object surface, while the Doppler shift caused by the sensor motion is approximately the same for all beams and is automatically subtracted from the measurements. This allows measurements of vibration fields of objects with high sensitivity from a moving platform. Scanning the linear array of laser beams in the transverse direction provides a two-dimensional vibration image of the surface. Performance of the sensor for ground vibration sensing for acoustic detection of buried objects has been investigated in the laboratory and field experiments. The sensor proved effective at detecting buried objects from a moving vehicle. [Work supported by the Office of Naval Research.]

**11:40**

**2aPAb2. Optical measurements of acoustic fields using refracto-vibrometry.** Thomas M. Huber (Phys., Gustavus Adolphus College, 800 W College Ave., Saint Peter, MN 56082, huber@gac.edu), Haley Anderson, Katelyn Espe, Ezekiel Haugen, Zane Michael, and Benjamin Rorem (Phys., Univ. of Michigan, Saint Peter, MN)

Refracto-vibrometry (RV) is an interferometric method for optically measuring acoustic fields. The measurement beam from a Polytec PSV-400 scanning laser Doppler vibrometer was directed through water or air 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 a large number of scan points, 2-D projections of propagating acoustic fields can be determined. Videos will be shown that enable visualization of acoustic wave propagation in air, water, and other transparent media. It is also possible to determine acoustic field distributions in 3-D through tomographic reconstruction of multiple RV projections through the acoustic field at different angles. It will be shown that there is strong agreement between the acoustic field distributions measured using a conventional needle hydrophone and the optical measurements using tomographic reconstruction of RV measurements. One of the limitations of the RV technique is that these optical measurements have a relatively low signal-to-noise ratio that may require averaging of a large number of ultrasound pulses. Techniques will be presented to minimize this acquisition time, including deep learning and frequency-sweep cross-correlation.

12:00

**2aPAb3. The use of optical methods for measuring weak acoustic shock waves in homogeneous air and close to reflecting boundaries.** Maria M. Karzova (Phys. Faculty, M. V. Lomonosov Moscow State Univ., Faculty of Phys., Leninskie Gory 1/2, Moscow 119991, Russian Federation, karzova@physics.msu.ru), Petr V. Yuldashev (Phys. Faculty, Lomonosov Moscow State Univ., Moscow, Russian Federation), Sebastien Ollivier (Laboratoire de Mécanique des Fluides et d'Acoustique, Unité Mixte de Recherche 5509, Université de Lyon, Ecole Centrale de Lyon, Université Claude Bernard Lyon I, Institut National des Sci. Appliquées de Lyon, Ctr. National de la Recherche Scientifique, Lyon, France), Vera Khokhlova (Phys. Faculty, Lomonosov Moscow State Univ., Moscow, Russian Federation), and Philippe Blanc-Benon (Laboratoire de Méanique des Fluides et d'Acoustique, Unité Mixte de Recherche 5509, Université de Lyon, Ecole Centrale de Lyon, Université Claude Bernard Lyon I, Institut National des Sci. Appliquées de Lyon, Ctr. National de la Recherche Scientifique, Ecully, France)

Measuring blast waves in laboratory-scale experiments using acoustical methods is a challenge. Optical methods provide an attractive possibility for recording pressure signatures of weak acoustic shocks. In this presentation, recent experiments for reconstructing pressure waveforms of spark-generated spherically divergent blast waves (1.8 kPa amplitude and 50 $\mu$s duration at 15 cm distance from the source) both in homogeneous air and close to reflecting surfaces are overviewed. Three methods were employed: shadowgraphy, schlieren, and interferometry. It was shown that shadowgraphy method allowed for measuring shock thickness and its amplitude. Schlieren technique provided reconstruction of the pressure waveforms in homogeneous air. In the reconstruction process the front geometry was assumed to be spherical or cylindrical. The exposure time of the high-speed camera was a limiting factor for the time resolution. Mach-Zehnder interferometer method was the most relevant for laboratory-scale measurements. The method reached 0.4 $\mu$s of time resolution, which was more than 6 times higher than that of 1/8-in condenser microphones. Moreover, the Mach-Zehnder interferometry allowed quantitative reconstruction of the pressure waveform without additional calibrations. The method was successfully applied for measuring waveforms in homogeneous air and reflected waves from rigid smooth and rough surfaces. [Work supported by ANR-10-LABX-0060/ANR-16-IDEX-0005.]

12:20

**2aPAb4. Acoustic velocimetry imaging.** Nathan E. Murray (National Ctr. for Physical Acoust., The Univ. of MS, 1 Coliseum Dr., University, MS 38677, nmurray@olemiss.edu), Charles E. Tinney (Appl. Res. Lab., U. of Texas at Austin, Austin, TX), and Gregory W. Lyons (Information Technol. Lab., U.S. Army Engineer Res. and Development Ctr., Vicksburg, MS)

We review a technique that we developed that combines shadowgraph or schlieren imaging with image processing tools to study the sound produced by turbulent shear flows. These flows are challenging acoustically because they are characterized by a distributed cloud of correlated sources that produce sound through multiple mechanisms. In the far field, 100 s of characteristic lengths from the source region, the pressure waves are purely acoustic, the apparent the source location shrinks to a point, and the common assumption is that the propagation behavior is spherical. In the near field, pressure signals from both acoustic and hydrodynamic (or evanescent) signatures are difficult to separate, especially when the convective Mach number of the sound-producing events is supersonic. Single-point measurements such as condenser-type microphones are blind to the direction of the approaching/departing pressure waves unless they are combined in arrays. Non-intrusive optical-based methods offer some benefit in this scenario and can be tailored in new and different ways. Examples include acoustic laser Doppler velocimetry and quantitative schlieren methods. Our technique combines modern digital camera technology (with high-resolution, high-speed capabilities) with tomographical image processing techniques images to yield a "picture" of the sound field that can provide quantitative assessment of the source distribution and source velocities. This optical technique is a strong complement for single-point or array microphone measurements.

## Session 2aPPa

# Psychological and Physiological Acoustics: Honoring William Yost's Contributions to Psychological Acoustics I

Robert A. Lutfi, Cochair
*Univ. of South Florida, Tampa, 4202 E. Fowler Avenue, Tampa, FL 33620*

Christopher A. Brown, Cochair
*Communication Science and Disorders, University of Pittsburgh, 4028 Forbes Tower, Pittsburgh, PA 15260*

**Chair's Introduction—9:30**

## *Invited Papers*



**9:35**

**2aPPa1. Does he ever sleep? Contributions to science and scientific administration.** Marjorie R. Leek (Res., Loma Linda VA Healthcare System, 11201 Benton St., Loma Linda, CA 92357, Marjorie.Leek@va.gov)

Bill Yost's contributions to organizations, institutions, and academic endeavors are myriad. Let us begin with his participation in the Acoustical Society of America—clearly the academic and professional home for most of us in this room. He has served this society in leadership roles for nearly fifty years, including stints as President and other positions in the ASA administration. His honors from the ASA are the most significant we offer – winning the Silver Medal for Psychological and Physiological Acoustics in 2006 and the Gold Medal in 2018. His contributions include hundreds of articles and books published in JASA and elsewhere, and he is an active participant in the national meetings of the Society. His academic career also includes important leadership roles in academic administrations at two major universities, serving, among other roles, as Dean, Research Administrator, and Department Chair. And all of this, while teaching courses, and writing and editing scholarly books and journal articles. As a leading auditory scientist, Yost enjoys multiple active collaborations world-wide, and has a compelling history of national grant funding. He is a legend and an aspirational example for his students and colleagues. But when does he sleep?

**9:55**

**2aPPa2. Brain imaging with iterated ripple noise.** Roy D. Patterson (Physiol., Development and Neurosci., Univ. of Cambridge, Downing St., Cambridge, Cambridgeshire CB2 3EG, United Kingdom, rdp1@cam.ac.uk)

In the early 1990's Bill Yost challenged me to test the Auditory Image Model (AIM) with Iterated Ripple Noise (IRN). Bill and Stan Sheft had established the mathematics of IRN using an autocorrelation model. We jointly demonstrated that the pitch algorithms in AIM and their autocorrelation model produced comparable estimates of pitch value and pitch strength. The cochlear simulations in the models both showed that IRNs with similar energy levels but different lags would produce similar spectral and temporal distributions of activity in the cochlea. This led to the intriguing idea that IRN could be used in brain imaging experiments to produce strong neural responses in centers all along the auditory pathway, and so make local imaging contrasts sensitive to aspects of activity beyond gross level, including pitch strength. Around 2000, Tim Griffith's group in London began to employ IRN melodies in PET and fMRI studies. More recently, Andre Rupp's group in Heidelberg has developed MEG imaging with IRN. These and other studies indicate that pitch processing begins in auditory cortex around Heschl's gyrus, bilaterally. These findings, based on Bill's informed use of IRN, are a great tribute to his extensive, inventive research record.

**10:15**

**2aPPa3. Going Yostal—Dr. Yost's conception of "retirement."** M. Torben Pastore (College of Health Solutions, Arizona State Univ., PO Box 870102, ASU, Tempe, AZ 852870102, m.torben.pastore@gmail.com)

Much will likely be made in this session of Dr. Yost's highly productive and often foundational scholarship, as well as his service to the field. This is as it should be, but Dr. Yost is currently as prolific and inquisitive a researcher as ever. This talk will offer an overview of our ongoing efforts in the Spatial Hearing Lab at Arizona State University to understand the size of the auditory scene, listeners' ability to direct their attention within auditory environments that contain more sources than can be perceived at a time, and, especially, the multi-sensory, multi-systems processing that integrates auditory spatial estimates into a multidimensional internal representation of the world around us.

**10:35**

**2aPPa4. Solving the cocktail party problem for bilateral cochlear implant users.** Christopher A. Brown (Commun. Sci. and Disord., Univ. of Pittsburgh, 4028 Forbes Tower, Pittsburgh, PA 15260, cbrown1@pitt.edu)

It is well-known that listeners with hearing impairment have trouble solving the cocktail party problem. Bilateral cochlear implant users have shown little or no perceptual abilities in this regard. This is due in large part to poor encoding of interaural time difference cues by these devices. Interaural level difference cues, which are well-encoded by bilateral cochlear implants, are not thought to facilitate benefit beyond improved signal-to-noise ratio benefits from head-shadow. An approach to mitigating this problem will be presented, that magnifies interaural level difference cues in real time to increase the perceptual distance between sound sources that are separated in the horizontal plane. The effects of ILD magnification on source localization and speech intelligibility in cocktail-party settings will be discussed.

TUESDAY MORNING, 8 DECEMBER 2020 — 11:15 A.M. TO 12:05 P.M.

# Session 2aPPb

## Psychological and Physiological Acoustics: Honoring William Yost's Contributions to Psychological Acoustics II

Robert A. Lutfi, Cochair
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Christopher A. Brown, Cochair
*Communication Science and Disorders, University of Pittsburgh, 4028 Forbes Tower, Pittsburgh, PA 15260*

**Chair's Introduction—11:15**

### *Invited Papers*

**11:20**

**2aPPb1. The precedence effect: Is it the same for interaural time and interaural level differences?** Raymond H. Dye (Psych., Loyola Univ. Chicago, 1032 W Sheridan Rd., Chicago, IL 60660-1537, rdye@luc.edu) and Sarah E. Darnell (Psych., Loyola Univ. Chicago, Chicago, IL)

Until recently, it was believed that the precedence effect operated for both interaural time and interaural level differences (ITDs and ILDs). Stecker and Brown (2010) demonstrated the absence of precedence effects for ILDs under conditions producing strong effects for ITDs. In order to provide a quantitative measure of the precedence effect, echo weights were measured using a correlational analysis as a function of echo delay for both ITDs and ILDs in 14 individuals for judgments based on binaural cues presented by the echo pulse. Stimuli were pairs of 3000-Hz 4-ms Gaussian pulses presented with echo delays between 4 and 96 ms. Echo weights at short echo delays (4 ms) were significantly higher for ILDs ($M = 0.445$) than ITDs ($M = 0.196$). Additional data were collected for conditions in which an ITD followed an ILD versus an ILD followed an ITD. Performance was far superior at short echo delays for ILDs following ITDs, with the leading ITD barely exerting any influence on subsequent processing of ILDs. We speculate that the mechanism by which ILDs are extracted (integration of sound pressure over time) makes later non-zero ILDs informative whereas later arriving ITDs provide little information about the location of sound sources.

11:40

**2aPPb2. Detection of modulated rippled noise.** Stanley Sheft (Dept. of Commun. Disord. and Sci., Rush Univ. Medical Ctr., 600 South Paulina St., 1015 AAC, Chicago, IL 60612, ssheft@gmail.com) and Valeriy Shafiro (Dept. of Commun. Disord. and Sci., Rush Univ. Medical Ctr., Chiago, IL)

Bill Yost's study of modulation detection interference (MDI) was in part motivated by his earlier work with temporally modulated rippled noise (RN) which indicated difficulty monitoring a single auditory channel in the presence of cross-channel modulation. The present work further evaluated parameters in the processing of dynamic spectral profiles. In the first experiment, gain in RN generation was sinusoidally modulated. If the gain sign remained unchanged during modulation, modulation resulted in a roughly constant increment in ripple-detection thresholds across change in modulation rate from 2 to 128 Hz. However, cross-sign gain modulation led to a lowpass result with thresholds increasing with modulation rate. A similar lowpass result was obtained in the second ripple-detection experiment in which delay in RN generation was modulated. The third experiment measured thresholds for detecting delay modulation. Thresholds, in terms of Weber ratios, rose with increasing modulation rate from 2 to 8 Hz. Across experiments, the lowpass results contrast with MDI in which interference decreases with increasing modulation rate. With dynamic spectral profiles, there is significant envelope modulation of individual auditory channels due to stimulus frequency excursions. Results suggest a limitation imposed by cross-channel envelope incoherence in utilizing this envelope modulation.

TUESDAY MORNING, 8 DECEMBER 2020 — 9:30 A.M. TO 11:00 A.M.



# Session 2aSAa

## Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics: Acoustic Metamaterials III

Christina J. Naify, Cochair
*Naval Research Lab, 4555 Overlook Ave. SW, Washington, D.C. 20375*

Alexey Titovich, Cochair
*Naval Undersea Warfare Ctr., Carderock Division, West Bethesda, MD 20817-5700*

Bogdan-Ioan Popa, Cochair
*Univ. of Michigan, Ann Arbor, MI 48109*

**Chair's Introduction—9:30**

### *Contributed Papers*

9:35

**2aSAa1. Metasurface-based sound diffusers for noise barrier applications.** Prachee Priyadarshinee (Mech. Eng., National Univ. of Singapore, #21-256B, 38 College Ave. East, Singapore 138601, Singapore, a0135595@u.nus.edu), Kian-Meng Lim, and Heow Pueh Lee (Mech. Eng., National Univ. of Singapore, Singapore, Singapore)

2-D sound diffusers installed with Helmholtz resonators have been investigated as top-edge noise barrier devices. The aim of this work was to study the effect of incorporating metasurfaces in sound diffusers on the broadband performance of noise barriers. This device can be exploited to use the astounding reflection and scattering properties of resonant metasurfaces and sound diffusers to achieve improvement in insertion loss. A three-dimensional indirect boundary element study was conducted to show promising improvement in noise barrier performance.

9:55

**2aSAa2. Scattering coefficients and calculation of acoustic fields in metamaterials.** Konstantin Dmitriev (Acoust., Moscow State Univ., Leninskie Gory, 1, 2, Moscow 119992, Russian Federation, presentatio@mail.ru) and Olga Rumyantseva (Acoust., Moscow State Univ., Moscow, Russian Federation)

An acoustic field in a metamaterial can be represented as the sum of an incident field and all fields scattered by metamaterial elements. Complex scattering coefficients are introduced to describe these scattered fields. The scattering coefficients satisfy the general relationships between their phase and amplitude. Set of their values on the complex plane is the circle with the center of -2i and the radius of 2 in the absence of absorption, or the inner part of this circle, if absorption exists. The use of the scattering coefficients makes it possible to take into account the processes of the multiple scattering between separate elements of the metamaterial, rather than between every two points of the medium, which is typical for traditional scattering problems. This fact simplifies calculation of the acoustic field for a wide class of metamaterials. The limited set of the possible values of the

scattering coefficients allows to consider solution of the inverse problem, i.e., to determine a design of a metamaterial element on base of the known scattering coefficients. However, the solution of this problem is unstable in a number of cases. [This study was supported by a grant from the Russian Science Foundation (Project No. 19-12-00098).]

**10:15**

**2aSAa3. Theoretical and experimental investigation of the transmission/reflection pressure fields of labyrinthine unit cells.** Abdelhalim Azbaid El Ouahabi (Project AURORA, School of Eng. & Informatics, Sussex Univ., Chichester 1 Bldg., Falmer, West Sussex BN1 9QJ, United Kingdom, A.Azbaid-El-Ouahabi@sussex.ac.uk) and Gianluca Memoli (Project AURORA, School of Eng. & Informatics, Sussex Univ., Brighton, United Kingdom)

Labyrinthine unit cells have been around for many years and have been central to the design of many metamaterial solutions. However, the literature does not present a reproducible analytical model to predict their behaviour both in transmission and reflection, thus limiting design optimisation in terms of bandwidth of operation and space occupied. In this work, we present an analytical model based on the transfer matrix method for the phase shift of the transmission/reflection labyrinthine unit cells. we validate our model using Finite Element Method (FEM) simulations—using a commercial software—and experimental measurements. We discuss the limitations of our approach and the perspectives opened by it.

**10:35**

**2aSAa4. Acoustic wave focusing using a 2.5-D graded metamaterial lens.** Yuanyan Zhao (Informatics, Univ. of Sussex, Falmer, Brighton BN1 9RH, United Kingdom, yz467@sussex.ac.uk), Sriram Subramanian (Comput. Sci., Univ. College London, London, United Kingdom), and Gianluca Memoli (Informatics, Univ. of Sussex, Brighton, United Kingdom)

The creation of a three-dimensional focal spot in air underpins applications in many fields. The literature presents many examples of focusing with acoustic metamaterials, but they are difficult to manufacture in large scale. Sonic crystals, are easier to make—since they are periodic arrangements of simple scatterers—but there are not many studies on achieving 3-D acoustic convergence through a 2-D sonic structure in air. In this study, we exploit the methods of sonic crystals to design the acoustic equivalent of a gradient index (GRIN) lens. We show that an extruded 2-D hexagonal lattice array of rigid cylinders with gradient diameters can be used to achieve the focusing of a 3-D shaped beam at audio frequencies i.e., a 2.5-D lens. We use finite-elements simulations with a commercial software to describe the device's performance in terms of band structure and equifrequency contours. We also present some preliminary measurements taken using a 3-D-printed model of our lens, designed for a focal length of 3.1 cm at 8 kHz. We find our 2.5-D GRIN lens works over a bandwidth of almost one octave. We discuss potential future uses, like in the correction of acoustical aberrations and in devices of flexible focal length.

TUESDAY MORNING, 8 DECEMBER 2020 — 11:15 A.M. TO 12:45 P.M.

## Session 2aSAb

## Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics: Acoustic Metamaterials IV

Christina J. Naify, Cochair
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**Chair's Introduction—11:15**

### *Contributed Papers*

**11:20**

**2aSAb1. Metamaterial-based acoustic camera enhancements.** Jonathan Eccles (School of Eng. and Informatics, Univ. of Sussex, Chichester 1, Falmer BN1 9QJ, United Kingdom, jpe24@sussex.ac.uk), Robert Cobden, Arash Pouryazdan, and Gianluca Memoli (School of Eng. and Informatics, Univ. of Sussex, Brighton, United Kingdom)

Acoustic cameras are a common approach to solving the challenge of sound localisation in non-destructive testing. The required microphone arrays, however, are often cumbersome: a large number of sensors is needed to achieve the required accuracy and their mutual distances need to be maximized to increase spatial resolution. Current acoustic cameras are not easily portable. In this work, we explore whether acoustic metamaterials can be used to create compact acoustic cameras, complete of autozoom objectives. We present our proof-of-concept prototype, comprising 8 MEMS microphones, using beamforming algorithms now sufficiently simple to run on a Raspberry Pi. We discuss how the performance changes when passing from a single lens to a two-lenses objective. Finally, we highlight the perspectives, the limitations and some key user scenarios for this approach.

11:40

**2aSAb2. A spatial sound delivery system for virtual and augmented reality.** Chinmay Rajguru (Eng. and Informatics, Univ. of Sussex, IBrighton BN1 9QJ, United Kingdom, c.rajguru@sussex.ac.uk), Arash Pouryazdan, and Gianluca Memoli (Eng. and Informatics, Univ. of Sussex, Brighton, United Kingdom)

Virtual and Augmented Reality (VR and AR) are increasingly becoming a consumer commodity, where sound plays a central role, usually delivered through headphones. Headphones, however, do not allow direct interactions between users outside the virtual world: other solutions may be necessary. In this work, we present a different approach: a sound projector, which can deliver spatial sound at a distance by forming a beam of audible sound. Our device consists of a portable speaker and two metamaterial lenses of different focal length, arranged in a telescope. We test our device by delivering signals between 5 and 6 kHz to moving people, tracked by a camera. We benchmark its performance with measurements and a pilot study, with human participants. Finally, we connect our projector to a VR headset and show how it can be used to deliver messages in VR. In this work, we discuss the limitations and perspectives of similar metamaterial-based delivery systems.

12:00

**2aSAb3. Internal acoustic energy of fluid scattering centers: Application to metamaterials.** Jose R. Alcaras (Phys., Univ. of Sao Paulo, Av. Bandeirantes, 3900, Ribeirao Preto, Sao Paulo 14040900, Brazil, jose.alcaras@usp.br) and Alexandre S. Martinez (Phys., Univ. of Sao Paulo, Ribeirao Preto, Sao Paulo, Brazil)

From its origins, scattering theory seeks to understand the way scattering patterns are related to scattering centers, in the far-field region. However, important information about the scattering object can be obtained via near and/or internal fields. Here, we calculate analytically the amount of energy stored inside fluid spheres and cylinders upon the single scattering of an incident monochromatic plane wave. First, we compute the internal scattering coefficients of the wave inside the scattering center, which can be a sphere or cylinder. Second, using this quantity, along with well established formulas, we calculate the potential and kinetic energy inside the scatterer. We expect to set up a method for theoretically obtaining standard parameters when measuring stability and structural nature of several types of scattering materials. Using scattering theory as means to study properties of materials, we expect to help further developing of acoustic metamaterials and improving the technological advances of this field.

12:20

**2aSAb4. Development and assessment of a low frequency acoustic liner design for landing systems noise minimization.** Frank Simon (DMPE, ONERA, 2 Ave. Edouard Belin, BP 74025, Toulouse 31055, France, frank.simon@onera.fr), Amin Ghouali (Safran Landing Systems, Vélizy, France), Valia Fascio (ATECA, Montauban, France), and Vincent Fleury (Dassault-Aviations, Saint-Cloud, France)

Acoustic liners for aeronautics have mostly a Helmholtz resonator behavior provided by perforated sheets backed by honeycombs. They may be used, for example, at the surface of a landing gear door to reduce the noise generated by a landing gear at landing. However, their acoustic absorption ability is naturally limited to medium and high frequencies due to thickness constraints. The LEONAR concept is proposed to overcome the problem of available space in case of "low frequency" range. The design consists in a meta-surface in which a perforated plate is coupled with tubes of variable lengths. This meta-surface covers a back cavity with limited volume and generate a significant shift in the frequency range of absorption, towards lower frequencies. First, an optimization problem is carried out to obtain the LEONAR meta-surface design that allows a maximal absorption coefficient between 400 and 1000 Hz, for a low thickness and a grazing flow up to Mach 0.2. Then the effect of impedance surface is checked by simulation of the radiated pressure field produced by a monopole source in presence of a landing gear door and compared with wind tunnel tests. The measurements showed significant attenuation within the prescribed range (D(OASPL) up to 1.5dBA).

## Session 2aSCa

## Speech Communication and Psychological and Physiological Acoustics: Reintroducing the High-Frequency Region to Speech Perception Research I

Ewa Jacewicz, Cochair
*Speech and Hearing Science, The Ohio State University, 1070 Carmack Road, 110 Pressey Hall, Columbus, OH 43210*

Robert A. Fox, Cochair
*Speech and Hearing Science, The Ohio State University, 1070 Carmack Road, 110 Pressey Hall, Columbus, OH 43210-1002*

**Chair's Introduction—9:30**

### *Invited Papers*

**9:35**

**2aSCa1. Using auditory electrophysiology testing to define the high frequencies across non-human mammal species.** Eric Bielefeld (The Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210, bielefeld.6@osu.edu) and J. R. DeBacker (The Ohio State Univ., Columbus, OH)

While the human's frequency range of audibility has been well documented at 20–20 000 Hz, the audible frequency ranges vary considerably from species to species. As such, that which constitutes the "high-frequency" range is quite different in mice, rats, guinea pigs, gerbils, chinchillas, and cats. These mammalian species have been frequently used in anatomical and physiological investigations of the peripheral and central auditory nervous systems. Much of what we know about the physiology of the auditory system is based upon on these animal species, yet many of their auditory systems operate in a much different frequency range from the human's. We will discuss the use of auditory electrophysiology to define the high-frequency ranges across several different mammal species, and then transition to a comparison of their high-frequency ranges to that of the human. Further, we will discuss the implications of trying to extrapolate insight into human auditory physiology from animal species whose high-frequency ranges are quite different from our own.

**9:55**

**2aSCa2. Benefits of extended high-frequency hearing for speech perception.** Brian B. Monson (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S Sixth St., Champaign, IL 61820, monson@illinois.edu)

Each species has a frequency range of hearing tailored to meet the demands placed upon that species by its environment and experiences. Why have humans retained extended high-frequency (EHF) hearing (i.e., sensitivity to acoustical energy beyond 8 kHz)? In this talk I will provide an overview of EHF hearing and report on studies from our lab revealing the utility of EHF hearing for speech perception. Historically, speech energy beyond 8 kHz was deemed unnecessary for applications of interest (i.e., telephony), and has often been neglected by tradition. We find, however, that speech energy beyond 13 kHz is audible for healthy young adult listeners. Access to EHFs improves (1) listeners' judgements of a talker's head orientation, and (2) speech recognition scores in a listening scenario where the target talker faces the listener, but masker talkers face away from the listener. This improvement is likely due to the directional nature of EHF energy radiating from the mouth and has some reliance on EHF spectral detail. Our results indicate that EHF energy in speech is audible and provides useful information about the speech signal.

**10:15**

**2aSCa3. Perception and use of high-frequency energy in defining indexical and segmental features in speech.** Jeremy Donai (Commun. Sci. and Disord., West Virginia Univ., 355 Oakland St., Morgantown, WV 26506, jeremy.donai@mail.wvu.edu) and Dwayne Paschall (Idexx Labs., Dallas, TX)

Historically, the focus of speech perception research has been on spectral energy below 4-5 kHz. Recent research has shown a substantial amount of useful perceptual information above this frequency cutoff. Our studies at the Auditory Perception Laboratory at West Virginia University have revealed usable information related to talker sex, talker identity, vowel identity, and listening effort for human and machine applications. These benefits have been found in quiet and noisy listening conditions. Current projects are examining the use of high-frequency energy (above 4 kHz) for automated recognition tasks in quiet and noise. Using various spectral and temporal features extracted from high-frequency energy from a large database of signals (approximately 7000 vowels), classification accuracy for vowel identity, speaker sex, and individual speaker identify has been found to be significantly above chance. Specific details of this project as well as additional projects will be included in this presentation. Implications of this line of research will also be discussed.

10:35

**2aSCa4. Information about talker dialect is available in high-frequency region.** Robert A. Fox (Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., 110 Pressey Hall, Columbus, OH 43210-1002, fox.2@osu.edu) and Ewa Jacewicz (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH)

Given that almost all cues to speech intelligibility are contained within the low-frequency region, high-pass filtering has not been used in speech perception research as often as low-pass filtering. We provide evidence from high-pass filtered speech that information about talker dialect is available in the high-frequency region even in the absence of intelligibility cues. Setting the upper frequency limit at 11 kHz, sentences edited out of spontaneous conversations of 20 talkers from Ohio and 20 from North Carolina were high-pass filtered with frequency cut-offs varying from 0.7 to 5.56 kHz and presented to listeners from Ohio. Results showed that listeners were still sensitive to differences between the two dialects at the two highest cut-offs, 3.32 kHz and 5.56 kHz, and dialect identification was mediated by talker sex. Also, identification of talker sex was affected by dialect, with Ohio variety providing more cues to talker sex at 3.32 kHz and the variety in North Carolina at 5.56 kHz. Speech intelligibility was reduced above 3 kHz, and Ohio speakers were more intelligible overall. Irrespective of the intelligibility loss, the results suggest that residual dialect cues are still distributed in the high frequency region, and are preserved differently in male and female voices.

TUESDAY MORNING, 8 DECEMBER 2020 — 11:15 A.M. TO 12:45 P.M.



## Session 2aSCb

## Speech Communication and Psychological and Physiological Acoustics: Reintroducing the High-Frequency Region to Speech Perception Research II

Ewa Jacewicz, Cochair
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Robert A. Fox, Cochair
*Speech and Hearing Science, The Ohio State University, 1070 Carmack Road, 110 Pressey Hall, Columbus, OH 43210-1002*

**Chair's Introduction—11:15**

### *Invited Papers*

11:20

**2aSCb1. Artificial bandwidth extension of speech signals.** Peter Vary (Communications Systems, RWTH Aachen Univ., Muffeter Weg 3a, Aachen 52074, Germany, vary@iks.rwth-aachen.de) and Thomas Schlien (Communications Systems, RWTH Aachen Univ., Aachen, Germany)

Artificial Bandwidth Extension (ABWE) algorithms try to reconstruct frequency components of speech signals which have been lost by coding/transmission. From a system theoretical point of view, this seems to be questionable. However, by exploiting redundancy of speech production as well as psychoacoustic properties of auditory perception, missing parts can subjectively be recovered or at least mimicked to a certain extent. The motivation for ABWE is to bridge the quality gap between the traditional narrowband telephony (200–3400 Hz) and the recent wideband services (*HD voice*, 50–7000 Hz). An HD terminal which is connected to a traditional phone, may activate ABWE for improving the audio quality and possibly the intelligibility of the received signal. In this contribution, state-of-the-art ABWE algorithms and coding standards including ABWE techniques are reviewed. ABWE algorithms are based on a source-filter model. The three main sub-tasks are to estimate the excitation signal, the spectral envelope and the time domain envelope, given either just the narrowband signal or in addition a few bits of side information. Furthermore, insights with respect to the relative importance of these three components are presented.

**11:40**

**2aSCb2. The importance of high-frequency hearing in a medico-legal context.** Brian C. Moore (Dept. of Psych., Univ. of Cambridge, Downing St., Cambridge CB2 3EB, United Kingdom, bcjm@cam.ac.uk)

People whose hearing is damaged by exposure to intense sounds, typically in noisy factories or during military service, often claim compensation. The most common complaint of such people is difficulty in understanding speech when background sounds are present. However, because direct measures of the intelligibility if speech in noise were perceived to be unreliable, compensation has traditionally been based on the audiogram. Exposure to intense sounds usually has its greatest effects on audiometric thresholds at 4, 6, and 8 kHz. However, in several countries, including the USA and the UK, compensation for occupational noise-induced hearing loss is usually calculated using the average of audiometric thresholds for selected frequencies up to 3 kHz, based on the implicit assumption that hearing loss for frequencies above 3 kHz has no material adverse consequences. In fact, several studies show that frequencies above 3 kHz are important for the perception of speech in background sounds and for sound localisation, especially for resolving front-back confusions. It is concluded that audiometric thresholds at 4 kHz and perhaps 6 kHz should be taken into account when assessing hearing in a medico-legal context. In addition, direct measures of the ability to understand speech in noise should be obtained.

**12:00**

**2aSCb3. Considerations for enhancing the perception of frequency-lowered speech.** Joshua M. Alexander (Speech, Lang., and Hearing Sci., Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, alexan14@purdue.edu)

Some frequency transposition and all frequency compression methods in today's hearing aids adhere to an untested assumption that spectral features of high-frequency speech cues, namely frication, need to be preserved or replicated after being lowered in frequency. Consequently, recommendations for optimizing the selection of parameters that control this signal-processing feature have focused on maximizing the amount of high-frequency energy moved into the aided audiogram or on maximizing the frequency separation between key speech contrasts after lowering. I will present evidence that suggests that preserving temporal modulation from the original speech at the auditory periphery should be a key consideration for the perception of frequency-lowered speech.

**12:20**

**2aSCb4. Digital enhancement of speech perception in noisy environments.** Peter Vary (Communications Systems, RWTH Aachen Univ., Muffeter Weg 3a, Aachen 52074, Germany, vary@iks.rwth-aachen.de) and Markus Niermann (Communications Systems, RWTH Aachen Univ., Aachen, Germany)

If speech is reproduced in an acoustically disturbed environment via a loudspeaker, the perception in terms of intelligibility or listening effort may be impaired. Typical situations arise in connection with mobile phones or public address systems. The location of the listener is called the *near-end* while the speech signal is received from the *far-end*. Usually, the acoustical background noise cannot be influenced, but speech perception may be enhanced by digital pre-processing of the loudspeaker signal. This pre-processing technique is called Near-End Listening Enhancement (NELE). The received signal is filtered adaptively, by taking the instantaneous characteristics of the background noise into account and by exploiting psychoacoustic properties of the auditory system such as the masking threshold. The increase of the loudspeaker volume is often not allowed or very limited, to avoid injury of hearing or damages of the loudspeaker and because of the listening comfort. In this contribution, state-of-the-art NELE algorithms operating in the short-term frequency domain are reviewed and novel time domain approaches are presented. Furthermore, the differing NELE constraints of public address systems and mobile phones are discussed.

## Session 2aSPa

# Signal Processing in Acoustics, Animal Bioacoustics, Engineering Acoustics, Underwater Acoustics, and Acoustical Oceanography: Acoustic Localization II

Zoi-Heleni Michalopoulou, Cochair
*Department of Mathematical Sciences, New Jersey Institute of Technology, 323 M. L. King Boulevard, Newark, NJ 07102*

Kainam T. Wong, Cochair
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Paul J. Gendron, Cochair
*ECE, Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747*

**Chair's Introduction—9:30**



### *Invited Papers*

**9:35**

**2aSPa1. Green's function retrieval and frequency-wavenumber methods for multiple source localization.** Max Denis (Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Washington, DC 20008, max_f_denis@hotmail.com), Sandra Collier, John M. Noble, W. C. K. Alberts, David A. Ligon, Leng K. Sim, and Deryck James (CCDC Army Res. Lab., Adelphi, MD)

In this work, Green's function retrieval and frequency-wavenumber methods are employed for multiple source localization in an outdoor environment. The Green's function retrieval methods are used to improve the signal-to-noise ratio of the beamforming maps. The frequency-wavenumber method is adapted to the plane-wave beamforming map to accurately localize real sources, while removing the appearance of ghost sources due to the data-association problem and mitigating missed detections for sources close together. Open field microphone array measurements of active and passive sources are investigated.

**9:55**

**2aSPa2. Sparse Bayesian learning for time-varying DOA estimation.** Yongsung Park (Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, yongsungpark@ucsd.edu), Florian Meyer, and Peter Gerstoft (Univ. of California, San Diego, La Jolla, CA)

Sparse Bayesian learning (SBL) provides sparse direction-of-arrival (DOA) estimation performance, and an SBL scheme with sequential processing is proposed for time-varying DOA estimation. SBL employs a Gaussian prior for the source signals and models variances of the source amplitudes as a hyperparameter. For sequential processing, the Gaussian-distributed source signals are modeled as Gaussian processes, and we consider the variances of the source amplitudes as the parameter of the covariance function in the Gaussian process. The sequential SBL estimates the variance parameter that evolves sequentially over time based on a state-space model. The suggested SBL with the sequential processing provides high-resolution capabilities for time-varying DOAs with varying source strengths or moving sources over time. The present method is evaluated by using simulated and real data (SWellEx-96 Event S5).

**10:15**

**2aSPa3. Linear and nonlinear Bayesian localization in ocean acoustics.** Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC V8W 2Y2, Canada, sdosso@uvic.ca)

Ocean-acoustic localization can be considered an inverse problem that involves estimating model parameters that specify the location of one or more acoustic sources and/or receivers based on fitting measured acoustic data, which can include observable quantities such as travel times, travel-time differences, modal dispersion, or acoustic-field structure. In a Bayesian inversion approach, data and prior information are used to compute the posterior probability density (PPD) of the model parameters, providing uncertainty analysis that quantifies the information content of the problem. The Bayesian formulation provides the generality to treat all uncertain parameters (e.g., both source and receiver locations, environmental properties, clock drifts) as unknowns subject to appropriate levels of prior information. Marginalizing over nuisance parameters can improve localization accuracy or at least account for parameter uncertainties in the localization uncertainty. Some Bayesian localization problems can be solved efficiently using linearization and iteration, with closed-form approximations for the PPD. In other cases, nonlinear (numerical) methods such as Markov-chain Monte Carlo sampling or trans-dimensional inversion are required. This talk will illustrate these concepts with a series of examples including array-element localization for moored and towed arrays, tracking autonomous underwater vehicles at a test range, marine-mammal localization, and multi-source matched-field localization

10:35

**2aSPa4. Effect of environmental uncertainty on source localization from mid-frequency tonals using convolutional neural networks.** Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., N251 ESC, Provo, UT 84602, tbn@byu.edu), Christian Escobar (Elec. and Comput. Eng., Univ. of Delaware, Newark, DE), Mason C. Acree (Phys. and Astronomy, Brigham Young Univ., Provo, UT), William Hodgkiss (Marine Physical Laboratory/Scripps Inst. of Oceanogr., San Diego, CA), David F. Van Komen (Phys. and Astronomy, Brigham Young Univ., Provo, UT), David P. Knobles (Phys., Knobles Sci. and Anal., Austin, TX), Mohsen Badiey, and Jhon A. Castro-Correa (Elec. and Comput. Eng., Univ. of Delaware, Newark, DE)

In ocean acoustics, the lack of labeled training data creates the need to train on simulated data. The simulated training data need to represent the environmental variability in the anticipated area where acoustic data are collected. Further, the manner in which environmental uncertainty should be incorporated is a significant question facing all ocean acoustics applications of machine and deep learning. To begin addressing this question, a case study is presented using mid-frequency tonal data recorded on a VLA. The time-varying tonal levels are input to a convolutional neural network (CNN) which finds the source range, depth, and speed and seabed type. The CNN is trained on simulated data and then tested on simulated data with different sound speed profiles and seabed types than used in training. The impact of this mismatch on the CNN predictions highlights the need to carefully account for environmental variability during training in order to provide robust machine and deep learning applications in ocean acoustics. [Work supported by the Office of Naval Research.]

TUESDAY MORNING, 8 DECEMBER 2020 — 9:30 A.M. TO 11:00 A.M.

## Session 2aSPb

## Signal Processing in Acoustics: Knowledge Discovery and Information Representation for Signal Processing in Acoustics I

Ananya Sen Gupta, Cochair
*Department of Electrical and Computer Engineering, University of Iowa, 103 S Capitol Street, Iowa City, IA 52242*

Benjamin N. Taft, Cochair
*Landmark Acoustics LLC, 1301 Cleveland Ave., Racine, WI 53405*

**Chair's Introduction—9:30**

### *Invited Papers*

9:35

**2aSPb1. Knowledge discovery from comparisons of specialized and conventional metrics determined from acoustic fields and measurements.** David R. Dowling (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109, drd@umich.edu), Brandon M. Lee (Mech. Eng., Univ. of Michigan, Ann Arbor, MI), Tyler J. Flynn (Mech. Eng., Univ. of Michigan, Laurel, MD), and Alexander S. Douglass (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Once acoustic signals (information streams) are recorded, there are myriad possible means for extracting and representing the embedded scientific or engineering knowledge that awaits discovery. Given the rich history of acoustic investigations, such knowledge extraction commonly involves comparisons based on conventional or specialized metrics chosen or devised to indicate critical underlying phenomena. Such comparisons may rely entirely on simulation results, experimental results, or a combination of the two. However, a suitable metric must be used in each case. This presentation provides four examples where conventional or specialized metric comparisons indicated new knowledge: (1) performing Monte-Carlo simulations and machine-learning predictions of the probability density function (PDF) of acoustic transmission loss (TL) in uncertain ocean sound channels to attain knowledge of how to best represent PDF(TL) when predicting it; (2) Using simulated acoustic-radiation data from vibrating structures and specialized dilation cross correlation metrics to remotely generate knowledge of experimental damage type and severity; (3) Comparing the measured horizontal coherence of acoustic fields and autoproducts to extract knowledge of their relative coherence lengths and bandwidths; and (4) Scaling of high-Reynolds-number hydrofoil-trailing-edge velocimetry and surface-pressure fluctuations measurements to attain knowledge of the near-wake conditions that lead to tonal noise. [Sponsored by ONR and NAVSEA.]

9:55

**2aSPb2. Interpreting the environmental information embodied in universal adaptive beamformers.** John R. Buck (ECE, Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, jbuck@umassd.edu) and Kathleen E. Wage (George Mason Univ., Fairfax, VA)

Practical adaptive beamformers often include regularization parameters to mitigate the impact of limited training data. These parameters may include the dominant subspace dimension for projection beamformers and the averaging window for the sample covariance matrix. Choosing the optimal value for these regularization parameters requires environmental knowledge not available to the algorithm in real time. Universal adaptive beamformers (UABFs) avoid the need for this knowledge. UABF outputs are a performance-weighted blend of the outputs of a competing family of beamformers. These blend weights embody operationalized knowledge about the array's environment. This environmental knowledge may be explicitly or implicitly represented, depending on the underlying beamformers within the universal framework. Previously proposed UABF implementations were universal over the dominant subspace dimension [Buck *et al.*, ASA (2018)] and the sample covariance matrix averaging window [Buck, ASA (2019)]. The former explicitly provides information about the number of interferers in the environment, while the latter implicitly captures information about the bearing rates of the interferers. Data analysis from an ocean acoustic experiment will demonstrate an example of the environmental information available from the blend weights. [Work funded by ONR 321US.]

## *Contributed Papers*

10:15

**2aSPb3. Performance weighted blended spectral estimation on experimental seaglider data.** Jeff Tucker (George Mason Univ., 4400 University Dr., Vienna, VA 22030, jtucke16@gmu.edu), Kathleen E. Wage (George Mason Univ., Fairfax, VA), and Lora Van Uffelen (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

A common problem in underwater acoustics is the non-parametric estimation of a power spectrum from time series data with windowed Fourier transforms. If characteristics of the environment such as signal to noise ratio or the frequency of loud line components are known then an appropriate window can be identified. When estimation is performed without a priori knowledge of the environment, an analyst often uses an ensemble of different windowed Fourier transforms and synthesizes the results. When no analyst is available this process must be automated. The performance weighted blended~(PWB) spectrum estimator automates the work of an analyst by weighting each estimator in an ensemble based on its performance and summing them to create a new estimator that performs as well or better than each estimator in the ensemble regardless of the characteristics of the environment [Tucker *et al.*, IEEE UASP Workshop (2019)]. This talk summarizes the PWB estimator, and compares it to existing estimators when applied to passive acoustic data collected from a hydrophone mounted on a Seaglider autonomous underwater vehicle. Additionally, the talk presents modifications to the original PWB estimator that offer improved robustness in the presence of loud transient signals, e.g., noise from glider hardware. [Work supported by ONR.]

10:35

**2aSPb4. An acoustic remote sensing method for high-precision speed estimation of micro-UUVs.** Kristen Railey (MIT/ WHOI, 77 Massachusetts Ave., Cambridge, MA 02139, krailey@mit.edu), Dino Dibiaso (Draper, Cambridge, MA), and Henrik Schmidt (MIT/ WHOI, Cambridge, MA)

Understanding the dominant sources of acoustic noise in unmanned underwater vehicles (UUVs) is important for passively tracking UUVs and for designing quieter propulsion systems. This work describes how the speed of a vehicle can be passively measured by the unique high frequency acoustic signature of a brushless DC motor propulsion system. First, the causes of high frequency tones were determined through direct measurements of two micro-UUVs and an isolated thruster. From this analysis, the vehicle noise was mapped to speed and the common and dominant features of noise were established: strong tones at the motor's pulse-width modulated frequency and its second harmonic, each of which is a carrier with sidebands at frequency intervals equivalent to the propeller rotation frequency multiplied by the poles of the motor. Field experiments were performed where the speed of two micro-UUVs was predicted by measuring the sidebands of the two dominant tones. When the mapping of rotational to translational speed of the UUV was known, this method produced speed predictions with 0.004 m/s accuracy. These findings are applicable to many off-the-shelf vehicles available today which rely on brushless DC motors and can be easily integrated into passive acoustic security systems for target motion analysis. [Work supported by the Department of Defense NDSEG program, ONR, and Draper.]

# Session 2aSPc

## Signal Processing in Acoustics, Animal Bioacoustics, Engineering Acoustics, Underwater Acoustics, and Acoustical Oceanography: Acoustic Localization III

Zoi-Heleni Michalopoulou, Cochair
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*School of General Engineering, Beihang University, New Main Building D-1107, 37 Xueyuan Road, Beijing 100083, China*

Paul J. Gendron, Cochair
*ECE, Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747*

**Chair's Introduction—11:15**

### *Invited Papers*

**11:20**

**2aSPc1. Perturbing sound speed field estimates to study quality of acoustic ranging.** Ashwin Sarma (BAE Systems/URI, BAE Systems/FAST Labs, 130 Daniel Webster Hwy. MER15-2350, Merrimack, RI 03054-4989, ashwin@ele.uri.edu), Rex Andrew (APL/UW, Seattle, WA), Geoffrey Edelson (BAE Systems, Merrimack, NH), and John Waterston (DARPA, Arlington, VA)

Acoustic ranging is clearly dependent on the sound speed field c(x,y,z), where the points (x,y,z) are constrained to the two-dimensional region defined by a submerged source at a certain point in the ocean and a submerged receiver at a different but unknown point in the ocean. An in-situ estimate of c(x,y,z) based solely on a sound speed profile taken at the source of known location and one taken at the receiver at a unknown location has been used to obtain reasonable ranging estimates over large source-receiver separation [Sarma *et al*., IEEE UASP (2019)]. We discuss an approach to perturb this estimate of c(x,y,z) in a physically sound manner consistent with internal wave and other ocean processes in order to study the effect on the accuracy of ranging methods. The approach can indeed be applied to any estimate of c(x,y,z). The ultimate motivation for this work is in-situ determination of the inherent variability expected for a specific range estimate from any range estimation technique.

**11:40**

**2aSPc2. Acoustic source localization in imperfectly known environments using frequency-differencing techniques.** David R. Dowling (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109, drd@umich.edu)

Remote unknown sound-source localization is a challenging task with applications in a variety of fields, such as underwater acoustics, atmospheric acoustics, structural acoustics, bio-medical ultrasound, animal bio-acoustics, and seismology. In nearly all cases, the localization algorithm is only successful when there is good cross correlation between remotely measured and predicted acoustic fields, a possibility that exits at low frequencies and/or short ranges in imperfectly known environments. However, at sufficiently high frequency and/or long range, the requisite measured-to-predicted-field correlation might never be high enough for successful source localization. However, recent investigations of the frequency-difference autoproduct, a quadratic product of two complex field amplitudes having different frequencies, suggest that it may have the phase structure of an acoustic field at the difference frequency. Thus, using sufficiently low difference frequencies, unknown sources may be localized at ranges where conventional techniques are unsuccessful by correlating measured and predicted (ideal) autoproducts. The underlying formulation of frequency-differencing techniques is presented along with examples drawn from simulations, laboratory experiments, and ocean propagation measurements that involve frequencies from fractions of a Hertz to more than 100 kHz, and propagation distances from tens of centimeters to hundreds of kilometers. [Sponsored by ONR, NAVSEA, and NSF.]

12:00

**2aSPc3. Automated two-dimensional localization of underwater acoustic transient impulses using vector sensor image processing.** Aaron M. Thode (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., MC 0206, La Jolla, CA 92093, athode@ucsd.edu), Alexander Conrad (Greeneridge Sci. Inc., Santa Barbara, CA), Emma Reeves Ozanich (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA), Rylan King (Naval Undersea Warfare Ctr., Newport, RI), Simon E. Freeman, Lauren A. Freeman (Naval Undersea Warfare Ctr., Middletown, RI), Brian Zgliczynski, Peter Gerstoft (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), and Katherine Kim (Greeneridge Sci. Inc., San Diego, CA)

An algorithm is presented for automatically localizing transient impulsive sounds collected on several autonomous underwater vector sensors, spaced 15 to 23 m apart. The procedure, which does not require precise time synchronization, exploits transient signals of interest arriving from different azimuthal directions on each sensor. For each sensor the method first constructs time-frequency representations of both the squared acoustic pressure (spectrogram) and dominant directionality of the active intensity (azigram). Within each azigram sets of time-frequency cells associated with transient energy arriving from a consistent azimuthal sector are identified. Standard image processing techniques then link sets that share similar duration and bandwidth between different sensors, after which the algorithm triangulates the source location using the azimuths associated with the detection set. Data collected from shallow coral reef environments demonstrate the algorithm's ability to detect SCUBA bubble plumes and humpback whale song, and reveal consistent spatial distributions of somniferous fish activity. Analytical estimates and direct evaluations both yield false transient localization rates of 3%–6% in the coral reef environment. Many localized pulses have low signal-to-noise ratios, whose distribution has a median of 7.7 dB and an IQR of 7.1 dB. [Work sponsored by DARPA PALS.]

## *Contributed Paper*

12:20

**2aSPc4. Blind localization of early room reflections from reverberant speech using phase aligned spatial correlation.** Tom Shlomo (Elec. and Comput. Eng., Ben Gurion Univ. of the Negev, Ben Gurion 1, Beer Sheva 8443944, Israel, tomshlomo@gmail.com) and Boaz Rafaely (Elec. and Comput. Eng., Ben Gurion Univ. of the Negev, Beer Sheva, Israel)

Blind estimation of the direction of arrival (DOA) and delay of room reflections from reverberant sound may be useful for a wide range of applications. However, the high temporal and spatial density of early room reflections limit existing methods to the detection of only a small number of reflections. This paper presents a novel method for blind estimation of the DOA and delay of early reflections that overcomes the limitations of existing solutions. The method is based on a signal model in which the reflection signals are explicitly modeled as delayed and scaled copies of the direct sound. A phase alignment transform of the spatial correlation matrices is proposed; this transform can separate reflections with different delays, enabling the detection and localization of reflections with similar DOAs. It is shown that the DOAs and delays of the early reflections can be estimated by separately analysing the left and right singular vectors of the transformed matrices. A simulation study of a speaker in a room recorded by a spherical array demonstrates the effectiveness of the proposed method for accurately localizing a large number of reflections.

## Session 2aSPd

# Signal Processing in Acoustics: Knowledge Discovery and Information Representation for Signal Processing in Acoustics II

Ananya Sen Gupta, Cochair
*Department of Electrical and Computer Engineering, University of Iowa, 103 S Capitol Street, Iowa City, IA 52242*

Benjamin N. Taft, Cochair
*Landmark Acoustics LLC, 1301 Cleveland Ave., Racine, WI 53405*

**Chair's Introduction—11:15**

## *Contributed Papers*

**11:20**

**2aSPd1. Utilization of airgun source signatures to improve seismic imaging of ocean water columns.** Zheguang Zou (Dept. of Phys. and Astronomy, Univ. of MS, 145 Hill Dr., University, MS 38677, zou@olemiss.edu) and Likun Zhang (Univ. of MS, Oxford, MS)

Seismic oceanography is a new interdiscipline which uses "legacy" marine seismic reflection data collected by the oil industry to image ocean water columns like eddies and internal waves. Unlike seafloor reflections, ocean water-column reflections are typically weak due to the low acoustic impedance contrast, which challenges seismic oceanography. Utilization of the acoustic characteristics of low-frequency, broadband airgun source can enhance water-column signals for better ocean seismic imaging. Acoustics characteristics of received direct-path, water-column, and seafloor signals are analyzed and applied to design adaptive filtering process to enhance water-column signals. The processing increases the signal-to-noise ratio of water-column signal and attenuates the direct waves without changing the spectral content of the airgun source. The results can be useful to improve seismic oceanography by providing better imaging of ocean water columns.

**11:40**

**2aSPd2. Determination of sperm whale orientation with respect to a single hydrophone.** George Drouant (Oregon Inst. of Technol., 3201 Campus Dr., Klamath Falls, OR 97601, george.drouant@oit.edu) and Juliette W. Ioup (Univ. of New Orleans, New Orleans, LA)

Sperm Whales (*Physter macrocephalus*) produce echolocation clicks composed of several pulses while searching for food when diving. An estimate of a whale's length can be calculated by determining the time interval between consecutive pulses in the echolocation click. That interval is the interpulse interval or IPI. We have developed methods to estimate the IPI of whales and thereby determine their lengths. The structure of a click from a sperm whale approaching a hydrophone is different from that of a whale moving away from a hydrophone (Zimmer, 2005). The methods we use to determine IPIs differ depending upon whether the clicks analyzed are from an approaching or leaving whale. Continuous Wavelet Transforms are used to determine if the clicks to be analyzed were produced by an approaching or leaving whale. The IPIs of data files from a mixture of approaching and leaving whales were analyzed by our method with good agreement with manual determinations of the IPIs.

## *Invited Papers*

**12:00**

**2aSPd3. Knowledge representation in goal-motivated decision making for intelligent active sonar.** Jill K. Nelson (Elec. and Comput. Eng., George Mason Univ., 4400 University Dr., MSN 1G5, Fairfax, VA 22030, jnelson@gmu.edu)

We consider knowledge representation in a goal-driven model for intelligent, or cognitive, active sonar designed to improve system performance and reduce sonar operator burden. The intelligent sonar uses observations of the environment (broadly defined) and predictions of future behavior to make decisions that leverage system resources to address a set of goals. These goals are created and managed by the system itself with possible input from the sonar operator. Decisions take the form of action selections, where actions may include transmit waveform, ping interval, illuminated region, etc. Each action is assigned a utility score, or metric, based on how well it addresses each system goal, with weighting according to how simultaneous (possibly competing) goals are prioritized by the system and/or operator. Metrics for evaluating candidate actions rely on signal processing models for localization and tracking in active sonar. To illustrate the impact of the structure of utility metrics on system behavior, we study the performance and decision-making behavior of the intelligent sonar system in simulated scenarios in which the system is tasked with addressing multiple search and track goals simultaneously. The effect of the chosen metric structure on the impact of goal prioritization and operator input is explored and discussed.

12:20

**2aSPd4. Cognitive sampling: The power of harnessing sampling strategies with machine learning techniques for enhanced machine classification and machine interpretation of acoustic datasets.** Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, 103 S Capitol St., Iowa City, IA 52242, ananya-sengupta@uiowa.edu)

Machine learning has received increased popularity in the acoustical community for its ability to interpret, classify and predict information autonomously from large-scale datasets. Beyond traditional supervised and semi-supervised learning architectures such as deep learning, and ensemble approaches, unsupervised learning networks such as autoencoders are being used to discover efficient data codings that enable hitherto unforeseen feature spaces. Unrelated yet deeply significant to the success of these myriad approaches is the spectral resolution of the signal itself as it has implications on the richness of information used by the learning architecture. Furthermore, much remains to be done in interpreting machine-learnt feature representations by domain experts and linking the domain knowledge to the machine-learnt knowledge. This talk will introduce a new way of enabling this link, by harnessing the power of compressive sampling and more generally, non-uniform sampling strategies and information theory to popular machine learning techniques. Specific applications will include a variety of acoustical applications involving spectral feature generation and interpretation.

TUESDAY MORNING, 8 DECEMBER 2020 9:30 A.M. TO 10:20 A.M.



## Session 2aUWa

## Underwater Acoustics: Array Processing in the Ocean

Derek Olson, Chair

*Oceanography, Naval Postgraduate School, Spanagel Hall, 833 Dyer Road, Monterey, CA 93943*

**Chair's Introduction—9:30**

### *Contributed Papers*

9:35

**2aUWa1. Low frequency synthetic aperture sonar: Spatial coherence analyse.** Fabien Novella (STIC, ENSTA Bretagne, 2 rue Francois Verny, Brest 29200, France, fabien.novella@ensta-bretagne.org), Yan Pailhas (NATO CMRE, La Spezia, Italy), Gilles Le Chenadec, and Isabelle Quidu (STIC, ENSTA Bretagne, Brest, France)

Spatial coherence is a widely studied notion for HF SAS. It allows the estimation of the navigation of the platform with an accuracy well below the wavelength of the system . It also allows, for high-frequency SAS images, the detection of stable reflectors generally symptomatic of manufactured objects. The theoretical background of spatial coherence is based on a derivation of the Van Cittert Zernike (VCZ) optics theorem adapted to acoustic. However, this theorem uses some assumptions (narrow band signals, Fresnel approximation, Born approximation and homogeneity of the scattering media) that are not necessary met for low frequency system. In this paper effects of wideband systems and presence of a target in the insonified volume are investigated. Do to so, the study is based on a simulation tool whose results are compared with real data from trials conducted by CMRE with a wideband LFSAS. Thanks to the use of a 2-D array, the two dimensional coherence can be analyzed. A sub-band study enables to respect the narrowband VCZ assumption. A statistical study on distribution of degree of coherence estimates allows to point out the presence of strong scatterers symptomatic of manufactured objects.

9:55

**2aUWa2. Vector sensor beam steering for underwater acoustic communications.** Fabricio A. Bozzi (University of Algarve, LARSyS, Ataíde de Oliveira, n77, 7D, Faro 8000-218, Portugal, fabriciobozzi@gmail.com) and Sérgio M. Jesus (University of Algarve, LARSyS, Faro, Portugal)

Acoustic Vector Sensors(VS) have been widely used for direction-of-arrival estimation in the past, while the employment of VS for underwater communications is a recent topic. Due to its compact size, VS may be used as a receiver in applications such as on board of Unmanned Underwater Vehicles (UUV), providing higher maneuverability and operational capabilities. Thus, in this study, the communication performance of a VS beam-steering technique is quantified. The performance analysis is made comparing the VS beam-steering with the versus time-reversal method in a shallow water simulation. The underwater channel is given by the seismo-acoustic propagation model, OASES. A Phase Shift Keying(PSK) modulation is adopted and the receiver includes a Decision Feedback Equalizer. An individual analysis of horizontal and vertical particle velocity is made to show the steering effects in the error performance. Moreover, as the steering angle is range-dependent, a comparison between methods is performed varying the range. The result indicates that the optimum steering angle, which brings less error, may not be related to the source direction. Furthermore, the proposed receiver shows the outperformance comparing to the time-reversal in both SNR and range, demonstrating the spatial filter advantage provided by this co-located sensor.

**10:15–10:20 Break**

## Session 2aUWb

# Underwater Acoustics: Seabed Acoustics

Derek Olson, Chair
*Oceanography, Naval Postgraduate School, Spanagel Hall, 833 Dyer Road, Monterey, CA 93943*

**Chair's Introduction—11:15**

## *Contributed Papers*

**11:20**

**2aUWb1. The connection between grain size, porosity, and Biot model parameters.** Nicholas P. Chotiros (Appl. Res. Labs., The Univ. of Texas at Austin, 86 Blenheim Crescent, Ruislip HA47HB, United Kingdom, chotiros@ieee.org)

The seabed is a porous acoustic medium, consisting of solid grains permeated by pore water. The Biot model for acoustic propagation in a porous medium is often regarded as a complex model that has an excessive number of input parameters, particularly permeability, pore size and tortuosity. Furthermore, the relationship between these parameters and the sediment classification, which is based on the mean grain size, is unclear. Borrowing from the geophysics and civil engineering communities, a relationship between grainsize and the Biot parameters was developed. This opens the way to predicting the acoustic properties of the seabed from geophysical seabed classifications. The relationship for well-sorted, unconsolidated sands and silts is well established. The transition from sand/silt to clay is where the problem becomes complicated for the following reasons: Because, in clay, a significant fraction of the fluid is attached to the solid platelets by electrostatic forces, and a significant proportion of the clay platelets may be suspended in the pore fluid, the boundary between pore water and skeletal frame needs to be clearly defined. The skeletal frame is sparse and supported by electrostatic forces, and therefore behaves differently to a mechanical packing of grains. [Work supported by ONR, Ocean Acoustics Program.]

**11:40**

**2aUWb2. Characterization of particle motion near offshore wind farm sites in the United States east coast.** Gopu R. Potty (Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg, Narragansett, RI 02882, potty@egr.uri.edu), James H. Miller (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Ying T. Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, RI), and Arthur E. Newhall (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

This study analyzes the particle motion data recorded on geophones and ocean bottom seismometers (OBXs) at various shallow water (water depth ~30 m) locations in the U.S. east coast near offshore wind farms. The Block Island, Rhode Island site was the location of the first offshore wind farm in the United States. Data were collected at this site during three different deployments in 2015, 2016, and 2017. Particle motion measurements were made during construction and operation of the wind turbines during these deployments. Data collected on a 3-axis geophone will be discussed in this study. This sensor package included a 3-axis geophone and a co-located hydrophone. This 3-axis geophone measured the particle motion (particle velocity) in three mutually perpendicular directions along with acoustic pressure. The same sensor package was also deployed ten miles east of Ocean City, Maryland and near the Virginia wind farm. During this deployment, in addition to the geophone package, four Ocean Bottom Recorders (OBXs) were also deployed as a line array to record construction noise. Data from these deployments will be presented and compared with published data from other windfarm sites. [Work supported by the U.S. Department of the Interior, Bureau of Ocean Energy Management (BOEM), Environmental Studies Program]

**12:00**

**2aUWb3. Impacts of range-dependent bottom type on mid-frequency propagation.** Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu) and Dajun Tang (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

The Target and Reverberation Experiment in 2013 (TREX13) focused on mid-frequency propagation and reverberation in shallow water off the coast of Panama City Beach, FL. The seafloor at the site primarily consisted of long-wavelength (150–200 m), small-amplitude (~1 m) sand ridges with narrow (20–30 m wide) regions of mud in the swales. Backscatter measurements using a multibeam sonar indicated that roughly 10%–20% of the seafloor was covered in mud. During the experiment, a source transmitting tones from 1.5 to 4.0 kHz was towed past two stationary vertical line arrays to measure the transmission loss (TL) along the track. The measured TL was found to be midway between the predictions for pure sand and pure mud sediments, despite the limited amount of mud at the site. Range-dependent propagation models which incorporate this variation in sediment type, indicate that the mud regions act as sinks of acoustic energy producing a significant increase in loss. Also, the spatial distribution is such that the mud regions cannot be discerned in the TL range-dependence. The implications of this range-dependence on both inversion and reverberation will be discussed. [Work supported by the U.S. Office of Naval Research.]

## Session 2pAAa

# Architectural Acoustics, Noise, and Structural Acoustics and Vibration: Sound Transmission and Impact Noise in Buildings III

Matthew Golden, Cochair
*Pliteq, 131 Royal Group Crecent, Woodbridge, L4H 1X9, Canada*

Benjamin M. Shafer, Cochair
*PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406*

**Chair's Introduction—1:05**

## *Invited Papers*



**1:10**

**2pAAa1. Long-term acoustical performance of underlayments under load.** Matthew Golden (Pliteq, 131 Royal Group Crecent, Woodbridge, ON L4H 1X9, Canada, mgolden@pliteq.com) and Musafere Faiz (Pliteq, Woodbridge, ON, Canada)

It is very common to use some sort of a resilient underlayment under gypsum-based topping products in multi-family construction to improve the airborne and impact acoustical performance. To prove acoustical performance, these products are tested in accordance to the relevant ASTM or ISO standards. An assembly is built and then tested in a short period of time. No evaluation of long-term performance is completed. This presentation will compare the performance of crumb rubber based GenieMat and several different plastic mesh entangled filament based underlayments in four key metrics. The first metric is the steady state creep under load. The second and third metrics are the change in performance in both load versus deflection and load versus natural frequency of the underlayments, pre and post steady state creep. The fourth metric is long term change in impact noise levels under load based on small scale testing similar to published standards.

**1:30**

**2pAAa2. An overview of new testing results on common wood joist and truss assemblies.** Michael Raley (Ecore Int.., 715 Fountain Ave., Lancaster, PA 17601, mike.raley@ecoreintl.com)

This presentation is an overview of new testing results for common 18 in. open-web truss, TJI, and $2 \times 10$ joist assemblies with gypsum concrete poured directly to the plywood subfloor. The presentation will discuss the effect of underlayment thickness, underlayment attachment (glued versus loose), and floor finish. It will also cover the difference in IIC and HIIC ratings for all tested iterations of the assembly, showing how the HIIC rating gives greater differentiation between flooring and underlayment combinations.

**1:50**

**2pAAa3. Sound transmission in mass timber floor-ceiling assemblies.** Evelyn Way (Maxxon Corp., 920 Hamel Rd., Hamel, MN 55340, evelynway@gmail.com), Erik Holmgreen (Maxxon Corp., Hamel, MN), Jason Ameen (Worthington Armstrong Ventures, Malvern, PA), and Ricky McLain (WoodWorks, Washington, DC)

As mass timber structures become more commonly used in the United States, laboratory acoustical data is needed to guide the design of these buildings. In the fall of 2019, a laboratory test program of airborne isolation per ASTM E90 and impact isolation per ASTM E492 of mass timber floor-ceiling assemblies was undertaken. Cross Laminated Timber (CLT), Dowel Laminated Timber (DLT) and Mass Plywood Panel (MPP) panel types were tested with ceiling conditions selected based on 2021 IBC fire code requirements. Comparisons will be made between panel types and other contemporary building types using broadband, low, and high frequency metrics.

*Contributed Paper*

2:10

**2pAAa4. Impact of the Chicago soundscape on building envelope design.** Anna C. Catton (Soundscape Eng., LLC, 729 W Ann Arbor Trl, Plymouth, MI 48170, acatton@soundscapeengineering.com) and Nathan Sevener (Soundscape Eng., LLC, Plymouth, MI)

Chicago's unique soundscape, dominated by buses and other street traffic, commuter trains, and the City's signature "L," both driven and augmented by the dense development in the urban core and preponderance of high-rise buildings, places demands on building envelope design. Building envelope design must address noise isolation at the façade of the structure and the frequent use of extensive glazing means that noise isolation has particularly large cost implications. Through measurements and acoustic modeling, the sound level can be quantified and predicted over the façade of a building under design. The results advise the envelope assemblies and acoustical design goals. This presentation will provide examples of the sound levels measured at various locations around the City, illustrate acoustic modeling that has been used to predict noise levels over the building façades under design, and describe envelope assemblies and acoustical design criteria that have been used.

TUESDAY AFTERNOON, 8 DECEMBER 2020 — 2:50 P.M. TO 3:40 P.M.

## Session 2pAAb

## Architectural Acoustics, Noise, and Structural Acoustics and Vibration: Sound Transmission and Impact Noise in Buildings IV

Matthew Golden, Cochair
*Pliteq, 131 Royal Group Crecent, Woodbridge, L4H 1X9, Canada*

Benjamin M. Shafer, Cochair
*PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406*

**Chair's Introduction—2:50**

*Contributed Papers*

2:55

**2pAAb1. An architectural investigation of mitigating noise to improve speech intelligibility in an open studio environment.** Christopher W. Kania (Architecture, Univ. of Florida, 135 West Central Boulevard, Ste. 500, Orlando, FL 32801, christopher.w.kania@gmail.com), Hassan Azad, and Martin Gold (Architecture, Univ. of Florida, Gainesville, FL)

This research project proposes a series of solutions to mitigate noise and improve speech intelligibility within an open architectural studio environment. Open studio environments pose a unique challenge compared to conventional open office environments. In addition to typical sources of noise found in open offices, such as conversation between co-workers, open studios contain noise generating equipment, such as plotters, 3-D printers and laser cutters. There is substantial research and evidence from the Finnish Institute of Occupational Health and AINS Group which conclude that different sources of noise in an open office environment hinder employee productivity. However, there is limited research that focuses on open architectural studio environments. Existing research regarding productivity, as it relates to acceptable noise level, will serve as a basis for application of Digitally Enhanced Sonic Enclosures (DESE) to mitigate noise within the open studio. Revit 2020, Rhinoceros 6 and Ease 4.4 will be utilized to analyze the sonic environment as it relates to noise and speech intelligibility in the CityLab Orlando design studio. Ultimately, this research offers an effective series of solutions, which can serve as architectural design guides to noise-related challenges in open studio environments.

3:15

**2pAAb2. What did Colonel Fabyan's acoustic anti-gravity device sound like?** Jon W. Mooney (Acoust. by JW Mooney, 418 11th St., De Witt, IA 52742, acoustics@jwmooney.com)

Somewhere within the Geneva (Illinois) Museum of History, next to an old movie theater popcorn machine and closed up behind a false wall, is hidden one of the last surviving components of Colonel Fabyan's acoustic anti-gravity device. Originally built by textile magnate, George Fabyan under the guidance of pseudo-science practitioners, it is Professor Wallace Sabine's examination and quick debunking of the device's anti-gravity powers that is credited with the redirection of Colonel Fabyan's fortune to the creation of Riverbank Acoustical Laboratories and the legitimate scientific study of acoustics. Although the levitation device is not currently on display, the museum's Curator of Collections & Exhibitions gave a small group of acousticians the unique opportunity in 2019 to examine the device's existing components and supporting archival files. Although no one in the group was interested in the device's flying capabilities, everyone was curious what the device must have sounded like during Sabine's testing. Reviewing newspaper accounts of the period, taking photographic measurements of the existing components, using physical and computer modeling, acoustic testing and simulation; we present animation and auralization of Colonel Fabyan's acoustic anti-gravity device in the lab and in-flight.

## Session 2pABa

# Animal Bioacoustics and Psychological and Physiological Acoustics: Celebrating Peter Narins' Contributions to Auditory Science III


Mark A. Bee, Cochair

*Ecology, Evolution & Behavior, University of Minnesota, 140 Gortner Laboratories, 1479 Gortner Ave., St. Paul, MN 55108*

Andrea M. Simmons, Cochair

*Cognitive, Linguistic, & Psychological Sciences, Brown University, 190 Thayer St., Box 1821, Providence, RI 02912-9067*


**Chair's Introduction—1:05**



### *Invited Papers*

**1:10**

**2pABa1. Visualization of the bullfrog middle ear.** Darlene Ketten (The Hearing Ctr., Boston Univ., Woods Hole Oceanographic Inst., Boston, MA 02215, dketten@whoi.edu) and Andrea M. Simmons (Cognit., Linguistic, & Psychol. Sci., Brown Univ., Providence, RI)

The structure and function of the anuran middle ear has been explored both physiologically and structurally. It is similar to that of some other vertebrates (birds and reptiles) in that it has a two-ossicle system consisting of an extracolumella and a columella. However, there are notable species and gender variations. Peter Narins has made significant contributions to our understanding of this unusual middle ear. One example is that the middle ear of terrestrial frogs has a cartilaginous operculum adjacent to the columellar footplate. Following on Peter's work, to better understand the middle ear *in situ*, we obtained Micro CT and Ultra High Resolution Helical (UHR CT) scans of the middle ear in intact heads of the bullfrog (*Rana catesbeiana)* on a Zeiss Xradia 520 Versa and a Siemens Volume Zoom at 10–100 $\mu$m isotropic voxel resolutions. Image analyses showed the core of the columella in this species is not a simple column but a twisted shaft. Further, the extracolumella is not constructed of dense cartilage but rather is a complex with exceptional soft, semi-cartilaginous tissues. Implications of these findings for middle ear function will be discussed.

**1:30**

**2pABa2. Good vibrations: The ground-shaking career of Peter Narins.** Jakob Christensen-Dalsgaard (Biology, Univ. of Southern Denmark, Campusvej 55, Odense DK-5230, Denmark, jcd@biology.sdu.dk)

Vibration signals are detected and used for communication by many different animals, and anurans—frogs and toads—are among the most sensitive. In this talk, I will review the current status of knowledge of anuran vibration sensitivity, largely based on the research by Peter Narins and collaborators. Although mechanical stimuli potentially can excite a variety of receptors, the most sensitive vibration receivers are found in the inner ear of anurans, specifically in the otolith organ sacculus and in the low-frequency hearing organ, the amphibian papilla, responding to whole-body vibrations at frequencies from 10 to 200 Hz. Substrate vibrations, in nature mostly generated as Rayleigh waves in the soil, are transmitted through the body to the inner ear, and the thresholds of the receptors to substrate acceleration are as low as 0.001 cm/s$^2$ in some species. This sensitivity is employed for various purposes by the anurans. A general use probably is warning against approaching predators, and some species have been reported to use vibrational communication. However, very few species of anurans have been studied extensively, and it is likely that vibrational communication is much more common, especially at close ranges, where vibration signals provide a relatively private information channel.

**1:50**

**2pABa3. Acoustic signals induce egg laying in túngara frogs.** Jenica L. Emerson, Karina Tom, Montana Pawek, Ariana Shulman, Brittany Watu, Caitlin Ha, Joshua Baek, Hong S. Lee (Biological Sci., Univ. of the Pacific, Stockton, CA), and Marcos Gridi-Papp (Biological Sci., Univ. of the Pacific, 3601 Pacific Ave., University of the Pacific, Stockton, CA 95211, mgridipapp@pacific.edu)

Male mating calls generally attract mates and deter competitors. In some vertebrates, however, mating calls can also influence ovarian development. Stimulation of the auditory system shifts the endocrine control of the female gonads but little is known about the underlying mechanisms. We studied túngara frogs because their mate choice responses to sound are well known. We stimulated isolated females with sound and monitored their egg laying to answer three questions: (1) Can egg laying be triggered by sound? (2) Is their response to sound specific? (3) Among conspecific male calls, are the most attractive ones also more likely to elicit egg laying? The results showed that egg laying in isolated females could be elicited by male advertisement calls in the presence of a pool of water. This response occurred within hours but it was abolished if the calls were modified. Among conspecific calls, complex ones known to have enhanced attractiveness to females did not differ significantly from simple ones in their ability to stimulate egg laying. The reproductive state response of the female to sound can therefore match or depart from her mate choice response indicating a complex neuroendocrine coordination with the auditory system.

**2:10**

**2pABa4. Determining vocal interactions and call timing in chorusing frogs: The contributions of Peter Narins.** Rama Ratnam (Biological and Life Sci., School of Arts and Sci., Ahmedabad Univ., Navrangpura, Ahmedabad, Gujarat 380009, India, rama.ratnam@ahduni.edu.in)

Numerous species of frogs call synchronously in a chorus by timing their calls to avoid call-overlap with conspecific neighbors. Although synchronous calling is most likely maintained by a call oscillator, several species of frogs including the Puerto Rican treefrog *Eleutherodactylus coqui* can adjust the timing of the oscillator period over a wide range so as to maintain call synchrony with competing stimuli. Here we describe some of the playback experiments performed by Peter Narins and his co-worker Randy Zelick on the timing capabilities of the oscillator in the coqui frog, particularly the ability to rapidly alter timing even when the silent window had a random onset and was short in duration. Narins recognized that characterizing calling behaviour in isolation, with synthetic stimuli, did not fully describe the acoustic challenges faced by calling males in natural assemblies. Consequently, Narins and his co-worker Jeffrey Brush obtained individual call timings from groups of two to five vocally interacting coqui frogs in a natural assembly. They showed that some males actively avoided call overlap with two, maybe three neighboring conspecifics, while some actively jammed a neighbor. The impact of these seminal studies, and the technical challenges that were overcome by Narins are discussed.

## Session 2pABb

# Animal Bioacoustics and Psychological and Physiological Acoustics: Celebrating Peter Narins' Contributions to Auditory Science IV

Mark A. Bee, Cochair
*Ecology, Evolution & Behavior, University of Minnesota, 140 Gortner Laboratories, 1479 Gortner Ave., St. Paul, MN 55108*

Andrea M. Simmons, Cochair
*Cognitive, Linguistic, & Psychological Sciences, Brown University, 190 Thayer St., Box 1821, Providence, RI 02912-9067*

**Chair's Introduction—2:50**



### *Invited Paper*

**2:55**

**2pABb1. Peter Narins impact on non-amphibian animal bioacoustics.** Arthur N. Popper (Univ. of Maryland, Biology/Psych. Bldg, College Park, MD 20742, apopper@umd.edu), Robert J. Dooling (Univ. of Maryland, College Park, MD), and Richard Fay (Loyola Univ. Chicago, Chicago, IL)

Combined, we have known Peter Narins for well over 120 years. We take great pleasure in honoring an "old" and greatly valued colleague and friend. We start by pointing out that none of us have done research with Peter, but that each off us knows, admires, and greatly values his work. For over 40 years, Peter's interdisciplinary blend of acoustics, psychophysics, physiology, and anatomy, often conducted in exotic places, has held us all in awe. Moreover, one of the most important part of Peter's work is that its impact extends far beyond amphibians. His work has helped us think about the bioacoustics of birds and fishes, and about the evolution of vertebrate hearing. His work has also stimulated research questions and how we understand research results for non-amphibian species. And, most importantly, it is not only the three of us who have benefitted from Peter's work – his contributions permeate all of animal bioacoustics and will do so for decades. Our talk will focus on how Peter's work has, and continues, to contribute to our own research, and that of many others. [Funded by three old guys.]

### *Contributed Paper*

**3:15**

**2pABb2. Peaks between peaks: Comparative intra-ear correlations in spontaneous otoacoustic emissions.** Christopher Bergevin (Phys. & Astronomy, York Univ., Petrie 240, 4700 Keele St., Toronto, ON M3J 1P3, Canada, cberge@yorku.ca), Tarnem Afify (Phys. & Astronomy, York Univ., Toronto, ON, Canada), and Rebecca Whiley (Biology, York Univ., Toronto, ON, Canada)

Spontaneous otoacoustic emissions (SOAEs) are a key facet of modern models of inner ear biomechanics, as the phenomenon provides crucial insight into the notion of the "active ear." Manifesting as an idiosyncratic array of spectral peaks unique to a given ear, individual SOAE peaks have nonstationary properties (e.g., amplitude and frequency modulations). Further, it has been demonstrated that interpeak relations between these "AM" and "FM" properties can be correlated, indicative of coupling of the underlying generation mechanisms. This study takes a comparative approach to characterizing these correlations in a wide variety of species exhibiting SOAEs (humans, birds, lizards) despite relatively disparate inner ear morphologies. Initial results are consistent with previous reports (e.g., van Dijk and Wit, 1990, 1998) in that SOAE interpeak correlations for a given ear are themselves idiosyncratic: Sometimes peaks (adjacent or not) exhibit correlated (positive or negative) AM and/or FM fluctuations with delays up to the order of milliseconds (typically longer for humans, shorter for lizards), while sometimes no correlation is observed. We explore implications for how such may constrain models treating the inner ear as a spatially distributed tonotopic system (e.g., various biophysical roles for coupling).

*Invited Paper*

**3:35**

**2pABb3. Acoustic discrimination of fine surface textures by echolocating free-tailed bats.** Michael Smotherman (Biology, Texas A&M Univ., 3258 TAMU, College Station, TX 77843-3258, smotherman@tamu.edu), Stephen Odunsi, Mikayla Hobbs, and Thomas Croft (Biology, Texas A&M Univ., College Station, TX)

Surface texture is an integral cue used by echolocating mammals for characterizing and forming a mental representation of an ensonified target. Bats need to be able to recognize and discriminate between different target surface textures. Previous work showed that bats rely on spectral cues embedded in echoes to resolve textures, but the resolution limits for this behavior are unknown. Mexican free-tailed bats (*Tadarida brasiliensis*) are fast, high-flying insectivorous bats that emit broadband FM multi-harmonic sonar pulses. We trained three bats to perform a two-alternative forced-choice assay in which they compared and selected the coarser of two sandpaper samples of different grit size. Commercial sandpaper grits decrease in mean particle size following an exponential function. We tested the bats ability to discriminate between 10 different grit sizes varying from 40 to 240 grit, corresponding to mean particles diameters varying from 425 to 50 $\mu$m. Bats discriminated all grits from a smooth plexiglass control and almost all grits from each other up to 180 versus 220 grit (82 vs 68 $\mu$m) but not 220 vs 240 (68 vs 54 $\mu$m), indicating an extraordinary minimum difference threshold of about 14 $\mu$m, which rivals human performance using tactile active sensing by finger touch.

TUESDAY AFTERNOON, 8 DECEMBER 2020 1:05 P.M. TO 2:35 P.M.

## Session 2pBAa

## Biomedical Acoustics, Signal Processing in Acoustics, Computational Acoustics, and Physical Acoustics: Modeling and Measuring Nonlinear Ultrasound Signals III

Thomas L. Szabo, Cochair
*Biomedical Engineering, Boston University, 44 Cummington Mall, Boston, MA 02215*

Keith A. Wear, Cochair
*Center for Devices and Radiological Health, Food and Drug Administration, Bldg 62 Room 2114, 10903 New Hampshire Ave., Silver Spring, MD 20993*

**Chair's Introduction—1:05**

*Invited Papers*

**1:10**

**2pBAa1. The nonlinear field produced by 1-D and 2-D imaging arrays with plane and diverging wave modes for ultrafast imaging.** Ting-Yu Lai (Bioengineering, Univ. of Washington, Seattle, WA) and Michalakis A. Averkiou (Bioengineering, Univ. of Washington, UW Box 355061, Seattle, WA 98195-5061, maverk@uw.edu)

Combining plane and diverging wave imaging (PWI/DWI) with tissue harmonic imaging (THI) may offer improvements in image quality in applications requiring very high frame rates (ultrafast). The beam shape and magnitude of the 2nd harmonic in tissue are important design considerations. We developed a numerical model based on the KZK equation and modeled three 1-D diagnostic arrays, L11-4v linear array, P4-2v phased array, and C5-2 convex array operating in PWI/DWI. Our numerical code predicts the nonlinear field of nonaxisymmetric sources and source configurations with elevation and no azimuthal foci (plane/diverging fields), which have not been modeled before. We showed that the second harmonic produced by ultrafast THI is 2-16 dB lower than that of focused beams for the imaging arrays considered when operated at the same maximum MI. This moderate difference of the second harmonic between PWI/DWI and focused ultrasound suggests that it is feasible to combine PWI/DWI and THI. We have also investigated harmonic generation produced by diagnostic 2-D arrays for 4-D THI. From our predictions for the second harmonic field we propose beamforming approaches for 4-D cardiac THI with focused ultrasound, PWI, and DWI that would result in frame rates of 10, 81, and 1275 Hz, respectively.

1:30

**2pBAa2. Keeping the grid size under control: Nyquist sampling in the Iterative nonlinear contrast source method.** Martin D. Verweij (Medical Imaging, Imaging Phys., Delft Univ. of Technol., Lorentzweg 1, Delft 2628 CD, The Netherlands, M.D.Verweij@tudelft.nl)

Nonlinear propagation of ultrasound is applied in medical imaging and therapy, as well as in nondestructive testing. Development of dedicated equipment and protocols, and interpretation of results, is facilitated by reliable simulation of nonlinear ultrasound fields. Typical applications require accurate representation of higher harmonics with amplitudes 120 dB below the fundamental, over spatio-temporal domains that span several hundreds of wavelengths/periods of the fundamental. Finite-element and finite-difference schemes usually require 10–20 points per wavelength/period of the highest harmonic, without a way to restrict the number of harmonics involved in the solution. Solving the nonlinear wave equation with these methods may therefore require a prohibitively large computational grid. Alternatively, the nonlinear wave problem may be cast into an integral equation, which can be solved iteratively. This is the basis of our ever-expanding Iterative Nonlinear Contrast Source method. Most importantly, this approach allows to explicitly limit the wavenumber/frequency range included in the solution, and hence enables sampling at the Nyquist rate set by the highest desired frequency. As a consequence, the computational grid can be relatively small without introducing aliasing. This presentation elucidates how this is achieved by an appropriate combination of numerical filtering and windowing operations during each iteration.

## *Contributed Paper*

1:50

**2pBAa3. Modeling of 3-D elastic waves in nonlinear elastic solids using the iterative nonlinear contrast source method.** Sundaraelangovan Selvam (Medical Imaging, Delft Univ. of Technol., Lorentzweg 1, Delft 2628CJ, The Netherlands, s.selvam@tudelft.nl), Arno Volker, Paul L. van Neer (Acoust. and Sonar, TNO, Hague, The Netherlands), Nico de Jong, and Martin D. Verweij (Medical Imaging, Delft Univ. of Technol., Delft, Netherlands)

In nondestructive testing, the generation of higher harmonics and the mixing of elastic waves are used to measure the nonlinearity parameters, which in turn are closely related to the microscopic state of the material. A 3-D numerical tool that can simulate large scale, four-dimensional, nonlinear elastic wavefields would be very useful for the reliable interpretation of experimental results. Here, a method is presented that can efficiently compute the nonlinear displacement fields in a homogeneous, isotropic elastic medium, taking into account its third-order elastic constants (A, B, C). The method is based on the Neumann iterative solution of an integral equation involving a Green's function of the linear 'background' medium, and a contrast source representing the nonlinearity of the medium. The integral equation is solved iteratively, and the computations are based on Fast Fourier Transforms using a sampling rate close to the Nyquist limit, i.e., two grid points of the shortest wavelength. The displacement fields are evaluated using the scalar and vector potential functions representing the compressional and shear displacements. In this presentation, the suitability of the INCS method for modeling the harmonics of 3-D nonlinear elastic waves and its validation by comparison with an analytical benchmark solution, will be presented.



## *Invited Paper*

2:10

**2pBAa4. Comparison between experimental and computational methods for the acoustic and thermal characterization of therapeutic ultrasound fields.** 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), Joshua Soneson (Johns Hopkins Univ. Appl. Phys. Labs., Silver Spring, MD), and Yunbo Liu (U.S. Food and Drug Administration, Silver Spring, MD)

For high intensity therapeutic ultrasound (HITU) devices, pre-clinical testing can include measurement of power, pressure/intensity and temperature distribution, acoustic and thermal simulations, and assessment of targeting accuracy and treatment monitoring. Though relevant standards are available, technical challenges remain because of the often focused, large amplitude pressure fields encountered. Measurement and modeling issues include using hydrophones and radiation force balances at HITU power levels, validation of simulation models, and a suitable tissue-mimicking material (TMM) for temperature measurements. To better understand these issues, a comparison study was undertaken between simulations and measurements of the HITU acoustic field distribution in water and TMM, and temperature rise in TMM. For the specific conditions of this study, the following results were obtained. In water, the simulated values for p+ and p- were 3% lower and 10% higher, respectively, than those measured by hydrophone. In TMM, the simulated values for p+ and p- were 2% and 10% higher than those measured by hydrophone, respectively. The simulated spatial-peak temporal-average intensity values in water and TMM were greater than those obtained by hydrophone by 3%. Simulated and measured end-of-sonication temperatures agreed to within their respective uncertainties (coefficients of variation of approximately 20% and 10%, respectively).

# Session 2pBAb

## Biomedical Acoustics: General Biomedical Acoustics: Backscatter I

Gianmarco Pinton, Chair
*Biomedical Engineering, University of North Carolina at Chapel Hill, 116 Manning Drive, Mary Ellen Jones Room 9212A, Chapel Hill, NC 27599*

**Chair's Introduction—1:05**

### *Contributed Papers*

**1:10**

**2pBAb1. Quantitative assessment of human cervix with correlation length ratio during pregnancy: An i*n vivo* longitudinal study.** Mohammadreza Kari (Medical Phys., UW Madison, 1122-F2 WIMR, 1111 Highland Ave., Madison, WI 53705, mkari@wisc.edu), Lindsey C. Carlson, Helen Feltovich (Maternal Fetal Medicine, Intermountain Healthcare, Provo, UT), and Timothy J. Hall (Medical Phys., UW Madison, Madison, WI)

We recently introduced a novel parameter called the correlation length ratio (CLR), a ratio of lateral to axial correlation lengths for backscattered echo signals, to detect the presence of elongated structures in soft tissues. Here, we look at variation of CLR and presence of elongated structures in human cervix during pregnancy using a longitudinal data from a group of women with normal pregnancy. Thirty women, ages ranging from 19 to 37 years, were scanned with ultrasound at five time points beginning at their normal first-trimester screening (8 –13 weeks) through term pregnancy (nominally 40 week). We computed correlation lengths, and CLR. To account for the system point spread function, an empirical cumulative distribution function of the CLR was obtained from a reference phantom. A one-sided threshold of 95% of the CDF was then computed. The difference between CLR in the sample and the threshold was computed and its variation with increasing gestational age was obtained. CLR estimation demonstrated a gradient along the length of the cervix. This longitudinal study also demonstrated an increase of about 0.5% (per week) in lateral correlation length and 0.8% increase in CLR with increasing gestational age, while axial correlation length was almost constant.

**1:30**

**2pBAb2. High-frequency quantitative ultrasound for characterizing collagen fiber alignment in murine tendon using angular dependence of integrated backscatter.** Sarah E. Wayson (Dept. of Biomedical Eng., Univ. of Rochester, 201 Robert B. Goergen Hall, Rochester, NY 14627, swayson@ur.rochester.edu), María Helguera (Tecnológico Mario Molina, Lagos de Moreno, Mexico), Denise C. Hocking (Dept. of Pharmacology and Physiol., Univ. of Rochester, Rochester, NY), and Diane Dalecki (Dept. of Biomedical Eng., Univ. of Rochester, Rochester, NY)

Two major complications following tendon surgery are adhesion formation and re-rupture. There is a need to develop an ultrasound imaging system for non-invasive, non-destructive, longitudinal monitoring of tendon structure throughout a rehabilitation protocol to optimize restoration of range of motion and mechanical properties. Type I collagen is the primary extracellular matrix protein in tendon, and its organization impacts tendon function. The objective of the present study is to develop a high-frequency quantitative ultrasound spectral analysis technique to characterize collagen fiber alignment in murine tendon. This work tests the hypothesis that the integrated backscatter coefficient (IBC) will exhibit anisotropy in murine tendon with aligned structure, and isotropy in murine liver with inhomogeneous structure. Backscattered echoes from murine tail tendon and liver were acquired at varying insonification angles using 38-MHz and 55-MHz single-element transducers. B-mode and IBC parametric images were computed, and the average IBC value in a region of interest was determined at each insonification angle. The IBC was angular-dependent in tendon and isotropic in liver. These data suggest that the IBC can be used to detect collagen fiber alignment in murine tendon, and contribute to establishing the foundation for a dedicated device to non-invasively monitor collagen remodeling during tendon healing.

**1:50**

**2pBAb3. Analysis of three-dimensional echo decorrelation and integrated backscatter imaging during *ex vivo* radiofrequency ablation.** Elmira Ghahramani Z. (Biomedical Eng., Univ. of Cincinnati3960 Cardiovascular Res. Center 231 Albert Sabin Way, Cincinnati, OH45267, ghahraea@mail.uc.edu), Peter D. Grimm, E. G. Sunethra Dayavansha, Kathryn Eary, Michael Swearengen, and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH)

Radiofrequency ablation experiments were performed in *ex vivo* bovine liver ($N = 14$) using a clinical ablation system (RITA 1500X generator and StarBurst probe, Angiodynamics). Three-dimensional (3-D) ultrasound images were acquired as complex beamformed echo volumes from a Siemens Acuson SC2000 scanner with a 4Z1c matrix array. Echo decorrelation images, depicting echo changes between sequential volumes (interframe time 18 ms), and integrated backscatter (IBS) images, depicting local changes in echogenicity relative to baseline, were computed in 3-D at 11 s intervals throughout each treatment. To assess potential for real-time prediction of tissue ablation, cumulative 3-D echo decorrelation and integrated backscatter images were compared to reconstructed ablation zones, segmented from optically scanned tissue sections. Both echo decorrelation and IBS provided good prediction of local ablation using receiver operator characteristic (ROC) curve analysis (area under curve 0.91 for echo decorrelation normalized to global echo energy; >0.85 for all parameters), with accuracy depending weakly on correlation window size. While ablation regions were predicted fairly well (Dice coefficients >0.6), ablation zone volume was not consistently predicted by thresholded parameter maps. Both echo decorrelation and IBS showed weak but statistically significant correlation with simultaneous, co-located tissue temperatures measured by four thermocouples on the RF probe.

2:10

**2pBAb4. Quantification of the backscatter coefficient in heterogeneous media: Comparison of attenuation compensation methods.** Laura Castañeda-Martinez (Instituto de Física, Universidad Nacional Autónoma de México, Circuito de la Investigacion S/N, Mexico City 04510, Mexico, lcastaneda@ciencias.unam.mx), Timothy J. Hall (Dept. of Medical Phys., Univ. of Wisconsin, Madison, Madison, WI), Noushin Jafarpisheh (Dept. of Elec. and Comput. Eng., Concordia Univ., Montreal, QC, Canada), Hayley Whitson (Dept. of Medical Phys., Univ. of Wisconsin, Madison, Madison, WI), Hassan Rivaz (Dept. of Elec. and Comput. Eng., Concordia Univ., Montreal, QC, Canada), and I. Rosado-Mendez (Instituto de Física, Universidad Nacional Autónoma de México, Mexico City, Mexico City, Mexico)

The estimation of backscatter coefficient, from *in vivo* tissue requires accurate compensation for intervening tissue attenuation. Current attenuation compensation methods (ACM) have been tested in fully or piecewise homogeneous phantoms. Thus, evidence of their performance in complex media remains scant. In this study we compare the performance of two ACM in a tissue-mimicking material with strong reflectors (SR). A gel-based phantom with SR was scanned with a L11-5v transducer on a Vantage 128 system (Verasonics). Estimates of from the phantom's background were obtained from the IQ echo signals using the known phantom attenuation (KA) or two ACM (a Constrained-Log-Difference (CLD) and a dynamic programming (DP) regularized method).The former method was used as gold standard. To analyze bias versus depth, a Pearson correlation coefficient was computed and the mean difference (MD) vs. depth was estimated. CLD-based showed a strong correlation ($r = 0.96$) with depth, while the correlation of DP-based estimates was weaker. DP-based values were closer to those obtained with KA (MD [0.9–1.15]) compared to the CLD-based estimates (MD [1.0–1.79]). The DP, regularized ACM show less sensitivity to the presence of SR compared to conventional ACM. We are currently using DP-based ACM to characterize breast lesions.



TUESDAY AFTERNOON, 8 DECEMBER 2020 — 2:50 P.M. TO 4:20 P.M.

## Session 2pBAc

## Biomedical Acoustics, Signal Processing in Acoustics, Computational Acoustics, and Physical Acoustics: Modelling and Measuring Nonlinear Ultrasound Signals IV

Thomas L. Szabo, Cochair
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Keith A. Wear, Cochair
*Center for Devices and Radiological Health, Food and Drug Administration, Bldg 62 Room 2114, 10903 New Hampshire Ave., Silver Spring, MD 20993*

**Chair's Introduction—2:50**

### *Invited Paper*

2:55

**2pBAc1. Construction of axisymmetric equivalent sources to facilitate the simulation of nonlinear acoustic fields in therapeutic ultrasound.** Wayne Kreider (CIMU / APL, Univ. of Washington, 1013 NE 40th St., Appl. Phys. Lab., Seattle, WA 98105, wkreider@uw.edu), Pavel B. Rosnitskiy, Petr V. Yuldashev, Oleg A. Sapozhnikov (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Tatiana D. Khokhlova (Div. of Gastroenterology, Univ. of Washington School of Medicine, Seattle, WA), Alex T. Peek (CIMU/APL, Univ. of Washington, Seattle, WA), and Vera Khokhlova (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

The development of new clinical applications of therapeutic ultrasound has been accompanied by efforts to standardize simulation methods for predicting nonlinear ultrasound fields. Although software tools for simulating nonlinear ultrasound propagation are increasingly available, computational burdens remain high, especially for fields characterized by strong focusing and significant harmonic content. Because nonlinear effects in such fields accumulate primarily over a short axial distance within the focal lobe, the nonlinear field can be approximated by considering an equivalent source with a similarly shaped focal lobe in the linearly focused beam. If an axisymmetric equivalent source can be identified, even strongly nonlinear fields can be accurately simulated with minimal computational burden using the Khokhlov-Zabolotskaya- Kuznetsov model. This approach has been implemented and validated in water for several real sources using only a simple set of linear field measurements to characterize the focal lobe of each source. To demonstrate the breadth of utility of this approach, holograms of non-axisymmetric fields distorted by phantoms were also used to define equivalent sources and estimate *in situ* nonlinear fields. Fast holography measurements combined with equivalent-source simulations can efficiently characterize the impact of different phantom geometries on nonlinear fields. [Work supported by NIH R01EB025187 and R01EB007643, and RSF 19-12-00148.]

## *Contributed Papers*

**3:15**

**2pBAc2. Spatial characterization of high intensity focused ultrasound fields in the brain.** Scott J. Schoen (Mech. Eng., Georgia Inst. of Technol., 901 Atlantic Dr. NW, Rm. 4125K, Atlanta, GA 30318, scottschoenjr@gatech.edu) and Costas Arvanitis (Dept. of Biomedical Eng., School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

High intensity focused ultrasound (HIFU) has seen widespread clinical adoption as a therapeutic tool, as it may be targeted noninvasively and without ionizing radiation. The converging sound waves induce thermal (e.g., ablation) and mechanical (e.g., radiation force) effects that may be localized to manipulate or destroy tissue. In addition to these primary effects, finite amplitude acoustic influences (e.g., second harmonic generation) become relevant due to the high pressure levels near the focal region. Specifically, improved localization of energy at sonic frequencies might be enabled by exploiting these effects. Here, through experimentally validated simulations, we demonstrate the relative importance of acoustical and material properties on the spatial and temporal properties of the resulting focal field, and the feasibility of attaining finite amplitude effects transcranially. While the relative amplitude of the second-order effect was found to scale with the source pressure and nonlinearity parameter, absorption and pulse modulation frequencies influenced both the distribution and spectral content of the field. These effects were observed for a water reference medium, and thus are likely to be even more prominent in tissue. Recent advancements in transskull focusing and understanding of the mechanical properties of neurons warrant further investigations, which will aim to explore these higher-order acoustic effects for noninvasive treatment of central nervous system diseases.

**3:35**

**2pBAc3. Transcranial cavitation localization by time difference of arrival algorithm using four sensors.** Zhongtao Hu (Dept. of Biomedical Eng., Washington Univ. in St. Louis, Radiation Oncology, Saint Louis, MO 63108, zhongtaohu@wustl.edu), Lu Xu, Chih-Yen Chien, Yaoheng Yang, Yan Gong (Dept. of Biomedical Eng., Washington Univ. in St. Louis, Saint Louis, MO), Dezhuang Ye (Mech. Eng. and Material Sci., Washington Univ. in St. Louis, Saint Louis, MO), Christopher P. Pacia (Dept. of Biomedical Eng., Washington Univ. in St. Louis, Saint Louis, MO), and Hong Chen (Washington Univ. in St.Louis, St Louis, MO)

Cavitation is widely existing in focused ultrasound (FUS)-mediated therapies in the brain, such as FUS in combination with microbubble-induced blood-brain barrier disruption, nonthermal ablation, as well as transcranial histotripsy therapy. Accurately knowing the 3-D location of cavitation in real time can improve the treatment targeting accuracy and avoid off-target tissue damage. However, the skull induces strong phase and amplitude aberration to the cavitation signals and presents a significant challenge to the transcranial cavitation localization. Existing techniques for 3-D cavitation localization use hemispherical multi-element arrays combined with passive beamforming and adaptive skull-specific correction algorithm. However, these techniques require expensive equipment and treatment planning. Their time-consuming computations limit applications in real-time cavitation monitoring, which is critically needed to ensure the safety and efficacy of the FUS treatment. The object of this study was to investigate the feasibility of using a four-sensor network to transcranially locate the cavitation source in 3-D by time difference of arrival algorithm. The positional error of transcranial cavitation localization with the human skull along x, y, and z axes were $1.7 \pm 1.2$ mm, $1.6 \pm 1.7$ mm, and $4.1 \pm 1.5$ mm, respectively. For comparison, the positional error of without the human skull were $1.2 \pm 1.8$ mm, $0.9 \pm 1.6$ mm, and $3.1 \pm 2.3$ mm, respectively.

**3:55**

**2pBAc4. Shock formation distance—A design parameter for high power acoustic energy transfer systems.** Vamsi C. Meesala (Virginia Tech, Blacksburg, VA 24061, vamsi24@vt.edu), Muhammad Hajj (Stevens Inst. of Technol., Hoboken, NJ), and Shima Shahab (Virginia Tech, Blacksburg, VA)

An interesting consequence of the nonlinear propagation of a finite-amplitude acoustic wave is the formation of a discontinuity in the pressure or also known as shock formation. It is associated with the dissipation of energy that is proportional to the cube of jump in the discontinuity. Such a loss in energy is detrimental for acoustic energy transfer applications as it will compromise the energy transfer efficiency. As such, the knowledge of the shock formation distance (SFD) is essential for designing efficient high-power energy transfer systems. We present an analytical frequency domain approach capable of predicting the SFD in the acoustic pressure distribution generated by a baffled disk with a general transverse deformation in a weakly viscous fluid medium. The nonlinear wave propagation is modeled using the Westervelt equation and the approach is based on solving it using the method of renormalization. The approach can be implemented either analytically or numerically to predict the SFD much faster than the time-domain simulations. The numerical implementation also allows the flexibility of using this approach for any source configuration. [This work was supported by NSF Grant No. ECCS-1711139, which is gratefully acknowledged.]

## Session 2pBAd

## Biomedical Acoustics: General Biomedical Acoustics: Backscatter II

Gianmarco Pinton, Chair
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**Chair's Introduction—2:50**

### *Contributed Papers*

**2:55**

**2pBAd1. Ultrasonic bone assessment: backscatter difference measurements of the femoral neck *in vivo*.** Kiera L. Downey (Dept. of Phys., Rhodes College, Nashville, TN, dowkl-22@rhodes.edu), Sarah I. Delahunt (Dept. of Phys., Rhodes College, Austin, TX), Loukas A. Georgiou (Dept. of Phys., Rhodes College, Memphis, TN), Aubrey J. Gray (Dept. of Phys., Rhodes College, Frederick, MD), Doni M. Thomas, Gia Pirro, Will R. Newman, Evan N. Main (Dept. of Phys., Rhodes College, Memphis, TN), Joshua T. Moore (Dept. of Phys., Rhodes College, Oxford, MS), and Brent K. Hoffmeister (Dept. of Phys., Rhodes College, Memphis, TN)

Introduction: Ultrasonic backscatter techniques are being developed to detect changes in bone caused by osteoporosis. The goal of this study was to evaluate the clinical utility of backscatter difference measurements at the femoral neck. Methods: Backscatter signals were acquired from the left and right femoral necks of 97 human volunteers using an ultrasonic imaging system (Terason T3000). The signals were analyzed to measure the normalized mean of the backscatter difference (nMBD), a quantity that represents the power difference between two portions of the same backscatter signal. Also, a bone sonometer (GE Achilles EXPII) was used to measure the stiffness index (SI) of the left and right heel bones. Results: Linear regression analysis was used to compare nMBD measurement at the femoral neck to SI measurements at the heel. A statistically significant ($R \geq 0.2$) correlation was observed between nMBD and SI. Conclusion: These results suggest that nMBD is sensitive to naturally occurring variations in bone tissue, and thus may be able to detect larger changes in bone caused by osteoporosis.

**3:15**

**2pBAd2. Study of ultrasonic scattering mechanisms in nonalcoholic fatty liver disease using liver histopathological slides.** Yashuo Wu (Bioacoustics Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 209 E University Ave. Apt. 221, Champaign, IL 61820, yashuow2@illinois.edu), Leonardo Lopez (Bioacoustics Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), Michael P. Andre, Rohit Loomba, Claude B. Sirlin (Dept. of Medicine, Univ. of California at San Diego, La Jolla, CA), Mark A. Valasek, Matthew A. Wallig (Dept. of Pathol., Univ. of Illinois at Urbana-Champaign, Urbana, IL), William D. O'Brien, and Aiguo Han (Bioacoustics Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Quantitative ultrasound (QUS) techniques are emerging to be useful for assessing nonalcoholic fatty liver disease (NAFLD), the accumulation of fat droplets in the liver without alcohol intake. Several QUS parameters, such as the backscatter coefficient (BSC), are known to be correlated with liver fat content. However, the underlining ultrasonic scattering mechanisms are not fully understood. Understanding these mechanisms may yield better QUS models for improved diagnostics. This study's purpose is to test the following hypothesized mechanism: hepatocyte nuclei are acoustic scattering sites and the accumulation of fat droplets changes the spatial distribution of hepatocyte nuclei, leading to the change of the structure function (SF), a component of the BSC. Hematoxylin and eosin-stained liver histopathological slides from 48 participants were digitized. Hepatocyte nuclei and fat droplets were automatically recognized. For each participant, the SF versus frequency was calculated from the nuclear distribution. The fat fraction (FF) was determined by the fractional surface area of fat droplets. The SF was positively correlated with the FF (Pearson's $r \sim 0.4$, $p < 10^{-4}$) between frequencies 1 and 30 MHz, supporting the hypothesized mechanism. Other potential mechanisms, such as the additional scattering caused by fat droplets, will also be discussed. [Work supported by R01DK106419 and R01CA226528-01A1.]

**3:35**

**2pBAd3. Ultrasonic measurements of the femoral neck in human volunteers using the backscatter amplitude decay constant.** Sarah I. Delahunt (Phys., Rhodes College, 206 Chesapeake Bay Ln. N, Austin, TX 78717, sarahidelahunt@gmail.com), Kiera L. Downey (Phys., Rhodes College, Nashville, TN), Loukas A. Georgiou (Phys., Rhodes College, Memphis, TN), Aubrey J. Gray (Phys., Rhodes College, Frederick, MD), Doni M. Thomas, Gia Pirro, Will R. Newman, Evan N. Main (Phys., Rhodes College, Memphis, TN), Joshua T. Moore (Phys., Rhodes College, Oxford, MS), and Brent K. Hoffmeister (Phys., Rhodes College, Memphis, TN)

Osteoporosis is a degenerative bone disease that affects millions of people worldwide. The goal of this study was to test a new ultrasonic technique developed for clinical bone assessment called the backscatter amplitude decay constant (BADC). Ultrasonic backscatter measurements were performed on 97 volunteers at the left and right femoral necks using an ultrasonic imaging system (Terason T3000) equipped with a 3.5 MHz convex array transducer. The backscatter signals were analyzed to determine the backscatter amplitude decay constant (BADC), a parameter that measures the exponential decay in the amplitude of the backscatter signal. For comparison, additional ultrasonic measurements were performed at the left and right heels using an ultrasonometer (GE Achilles EXPII) to measure the stiffness index of the calcaneus. BADC demonstrated weak but statistically significant correlations with stiffness index ($R < 0.25$, $p < 0.05$). With further refinement of the measurement technique, BADC may be a useful parameter for ultrasonic bone assessment.

**3:55**

**2pBAd4. New views of tissue scattering.** Kevin J. Parker (Elec. & Comput. Eng., Univ. of Rochester, Comput. Studies Bldg. 724, Box 270231, Rochester, NY 14627-0231, kevin.parker@rochester.edu)

What causes scattering of ultrasound from normal soft tissues such as the liver, thyroid, and prostate? Commonly, the answer is formulated around the properties of spherical scatterers, related to cellular shapes and sizes. However an alternative view is that the closely packed cells forming the

tissue parenchyma create the reference media, and the long cylindrical-shaped fluid vessels serve as the scattering sites. Under a weak scattering or Born approximation for the extracellular fluid in the vessels, and assuming an isotropic distribution of cylindrical channels across a wide range of diameters, consistent with a fractal branching pattern, some theoretical predictions can be made. Our model predicts that backscatter increases as a power law of frequency, where the power law is determined by the fractal dimension. These results are consistent with the pioneering measurements of Campbell and Waag. Furthermore, the normalized histogram of echo amplitudes is found to be related to the classical Burr distribution, with the key power law parameter directly related to the fractal dimension, expected in the range of 2 to 3 for normal vasculature. Thus, the first and second order statistics of backscatter from soft vascularized tissues appear to be determined by fractal branching cylindrical vessels.

TUESDAY AFTERNOON, 8 DECEMBER 2020 1:05 P.M. TO 2:35 P.M.

## Session 2pCA

# Computational Acoustics, Physical Acoustics, Signal Processing in Acoustics, and Underwater Acoustics: Ray Methods Across Acoustics III

Michelle E. Swearingen, Cochair
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Jennifer Cooper, Cochair
*Applied Physics Laboratory, Johns Hopkins University, 11100 Johns Hopkins Rd., Mailstop 8-220, Laurel, MD 20723*

**Chair's Introduction—1:05**

### *Invited Paper*

**1:10**

**2pCA1. A parametric study of long-range atmospheric sound propagation using Bellhop Ray-tracing Model.** Hammad Hussain (Electron. System, Norwegian Univ. of Sci. and Tecjmology, NTNU, Trondheim 7491, Norway, hammad.hussain@ntnu.no) and Guillaume Dutilleux (Electron. System, Norwegian Univ. of Sci. and Tecjmology, Trondheim, Norway)

In the context of research on the measurement of long-range attenuation of noise from terrestrial sound sources, a parametric study of atmospheric sound propagation channel characteristics as a function of source height, ground characteristics and meteorological conditions is presented. The study relies on ray tracing. The Bellhop ray-tracing model which is well known in underwater acoustics has been used here. In this paper, the accuracy of Bellhop's predictions in the atmosphere is first addressed by comparison with published benchmarks cases by Attenborough et al and results from Salomon's ray model. No significant discrepancy was noticed with respect to these references. The second part of the paper presents a parametric study for source heights ranging from 0.05 m to 200 m, a grid of receivers at ranges between 200 m and 2 km from the source and between 2 and 50 m height. A homogeneous flat absorbing ground described by a complex reflection factor is assumed. For the atmospheric conditions, a subset of the WiSi classification was considered. The results are analyzed from the point of view of the receiver and discussed in terms of attenuation, number of arrivals and number of reflections from the ground.

### *Contributed Papers*

**1:30**

**2pCA2. Wavefront tracing for undersea acoustic propagation.** David J. Pate (Sensors and Electromagnetic Applications Lab., Georgia Inst. of Technol., 7220 Richardson Rd., Smyrna, GA 30080, david.pate@gtri.gatech.edu)

An acoustic wavefront tracing algorithm and corresponding computational implementation is presented. Similar to ray tracing and beam tracing, wavefront tracing casts independent rays that refract through the medium and reflect from boundaries. However, with wavefront tracing these rays step congruously and are connected as a triangulated surface. As this triangulated surface moves through space and time it contacts boundaries, receivers, or other points of interest, thereby providing all possible paths from the source. As each triangle steps forward, three space-filling tetrahedra are formed and acoustic properties are then linearly interpolated via barycentric coordinates. This interpolation also applies at a boundary surface, allowing for both specular reflection and diffuse scattering. The algorithm is derived from the wavefront *construction* method, in which the triangulated surface is adaptively modified to maintain a desired level of detail. However, by keeping the wavefront topology fixed, along with an efficient encoding of the algorithm, highly detailed simulations with millions of triangles are achievable. Borrowing from the computer graphics community, a bounding volume hierarchy is used to accelerate the computation of ray–boundary intersections. An example simulation of a synthetic aperture sonar ping is provided, in which the wavefront comprises 21 million triangles and a rippled seafloor comprises five hundred thousand triangles.

**1:50**

**2pCA3. Modeling underwater propagation of acoustic vortex beams by ray methods.** Zheguang Zou (Dept. of Phys. and Astronomy, Univ. of MS, 145 Hill Dr., University, MS 38677, zou@olemiss.edu), Xudong Fan (Dept. of Phys. and Astronomy, Univ. of MS, University, MS), and Likun Zhang (Dept. of Phys. and Astronomy, Univ. of MS, Oxford, MS)

Acoustic vortex beams have application potentials in the fields of object manipulation and high-speed communication. Due to the additional azimuthal features, modeling the propagation of vortex beams usually requires three-dimensional (3-D) models, which are of high computational cost, especially for large-scale simulations. Here, the two-dimensional (2-D) BELLHOP ray-tracing model is used to simulate the 3-D propagation of acoustic vortex beams in the deep, long-range ocean environment for a low computational cost. The vortex source is constructed by multiple point sources, each with a predefined initial phase to form the vortex wavefront. The 3-D vortex acoustic field is modeled by a sum of the fields simulated for the individual sources in 2-D environments. The results reveal that the traditional 2-D ray-tracing model is an effective tool to study vortex beam propagation in inhomogeneous media, with less complexity compared to finite element simulations using COMSOL Multiphysics.

**2:10**

**2pCA4. Ray tracing for modeling underwater acoustic communications channel between moving platforms, with moving ocean surface and changing bathymetry.** Paul Hursky (Sonar-synesthetics, 4825 Fairport Way, San Diego, CA 92130, paul.hursky@gmail.com)

Underwater acoustic communication systems must contend with a time-varying channel due to motion of source, receiver, point of reflection from ocean surface waves, and point of reflection from the seabed over changing bathymetry. These motions produce Doppler effects, which impact acomms systems. The signal processing to equalize such channel fading typically consists of a number of tracking loops, whose testing requires time varying realizations of such channels. A realistic channel model would enable improved modem designs and hardware-in-the-loop testing. We will focus on how to handle the ocean surface and the changing bathymetry. We propose an architecture which pre-computes the ray tracing for the part of the ocean that is not changing, then updates only small subsets of the ocean where dynamics occur. This consists of identifying the point of specular reflection, then interpolating the source-to-reflector and receiver-to-reflector rays to produce the union of these two paths. Such acoustic paths may be shadowed by the shape of the ocean surface. We will present examples of a time-varying channel impulse response, using the combined travel times, and discuss how dynamic ray tracing can be used to produce the amplitude of the joined paths.



TUESDAY AFTERNOON, 8 DECEMBER 2020 — 1:05 P.M. TO 2:35 P.M.

## Session 2pMUa

## Musical Acoustics and Education in Acoustics: Musical Acoustics Education at the Undergraduate Level II

Andrew C. Morrison, Chair
*Natural Science, Joliet Junior College, 1215 Houbolt Dr., Joliet, IL 60431*

**Chair's Introduction—1:05**

### *Invited Papers*

**1:10**

**2pMUa1. An acoustic impedance probe for the teaching of musical acoustics to non-majors.** Herbert Jaeger (Phys., Miami Univ., 500 E Spring St., Oxford, OH 45056, jaegerh@MiamiOH.edu)

Physics of Music or similar titles are found in the course catalogues of many colleges and universities. Almost always such courses are directed at non-physics majors and even non-science majors. As a result the course materials must be presented without reliance on advanced mathematical concepts, and thus a great deal of in-class demonstrations combined with video clips are used instead. For example, the concept of acoustic impedance, when introduced rigorously, requires the use of complex algebra. However, a discussion on a qualitative level can be done in a way that is accessible to undergraduate students without a great deal of math background. In order to introduce some measure of quantitative discussion, we are using a simple home-built impedance probe to record the input impedances of simple air columns as well as real instruments. In this talk we will discuss the details of the impedance transducer and show how it can be used to demonstrate how a flared bell changes the resonances of cylindrical air columns, the effect of differently shaped mouth pieces on the acoustic properties of a brass instrument, and a number of other characteristics of musical instruments.

1:30

**2pMUa2. Laboratory measurements of conical reed woodwinds in a *Physics of Music* class.** Randy Worland (Phys., Univ. of Puget Sound, 1500 N. Warner, Tacoma, WA 98416, worland@pugetsound.edu)

In our *Physics of Music* class for non-science majors, wind instruments are initially modeled by studying resonances of open and closed (i.e., one end closed) cylindrical tubes. Among the woodwinds, flutes and clarinets fit the open and closed cylinder models quite well. However, the acoustical properties of closed (i.e., reed driven) conical woodwinds, such as oboes, bassoons, and saxophones, are less intuitive and more difficult to motivate in a course for non-science majors. The laboratory exercise described here employs a straightforward analysis of simple dimensional measurements of bore profile and tone-hole location to verify a few of these properties, and to illustrate some of the acoustical differences between flutes, clarinets, and conical woodwinds. For pedagogical reasons described in this presentation, a straight soprano saxophone provides an ideal prototype for this study, although more commonly available alto and tenor saxes can also be used.

## *Contributed Paper*

1:50

**2pMUa3. Converting performance spaces into lecture halls.** Erin Beaudoin (Phys., Whitman College, Walla Walla, WA) and Kurt R. Hoffman (Phys., Whitman College, 345 Boyer Ave., Hall of Sci., Walla Walla, WA 99362, hoffman@whitman.edu)

Identifying classroom spaces able to accommodate students consistent with social distancing guidelines is a major logistical concern for Academic institutions attempting to have in-person instruction during the pandemic. The solution to this problem is to adapt larger campus spaces such as gymnasiums, public spaces, or performance venues, for classroom usage. This work focuses on measuring important acoustical parameters in these planned lecture spaces such as reverberation time (T20), speech clarity (C50), background noise decibel level, and sound pressure level differences in a grid pattern in the room. We studied existing classrooms as well as the target spaces for comparison and to optimize the parameter values for the new lecture spaces. Acoustic measurements were taken and analyzed using a sine sweep method with a calibrated microphone. The T20 of the rooms were calculated using a Sabine model to compare with the measured values and to guide plans for reducing T20 to transform the spaces to host lectures and discussion. Because the gymnasiums will also revert to the normal function, the proposed solutions must be temporary and easily added or removed.

## *Invited Paper*

2:10

**2pMUa4. Solving real-world problems in a physics of music class.** Jack A. Dostal (Dept. of Phys., Wake Forest Univ., PO Box 7507, Winston Salem, NC 27109, dostalja@wfu.edu)

The Physics of Music is a general education science class open to all students at Wake Forest University. Topics covered include wave physics, hearing, voice/singing, musical instrument function and performance (winds, strings, and percussion), room acoustics, and more. We have found great value in applying the concepts learned in class to practical, real-world problems. In the past offerings of the course, students worked on two projects involving Wake Forest's Department of Theatre (musical instruments) and Financial Aid Office (room acoustics). Students in the course communicated in person with students, faculty, and staff in these areas. They later used what they learned in class and discussions to construct equipment for use by the department in question. In this talk, I will describe how these two projects were incorporated into the laboratory portion of the course as well as how connections can be made to uncover other real-world opportunities.

## Session 2pMUb

# Musical Acoustics and Education in Acoustics: Musical Acoustics Education at the Undergraduate Level III

Andrew C. Morrison, Chair
*Natural Science, Joliet Junior College, 1215 Houbolt Dr., Joliet, IL 60431*

**Chair's Introduction—2:50**

### *Invited Paper*

**2:55**

**2pMUb1. Optimal valve slide length for brass instruments.** Frederick J. Young (Elec. Eng., Carnegie Mellon Univ., Carnegie Inst. of Technol., Bradford, PA 16701, youngfj@youngbros.com)

One of the first patents for compensating valve brass instruments by US Letters Patent #457,337 dated 11 August 1891 of Fontaine Besson is studied and optimized to obtain a much better in tune compensating instrument. The variance of intonation for the Besson instrument is 18.86%. An optimized four valve brass instruments is devised that lowers the variance to 4.22%. However, the first valve is flat by 7.39% on both sides of the instrument. Because intonation errors of ±5% are tolerable a spring loaded trigger must be provided to shorten the first valve when used alone. An EXCEL spreadsheet is presented analyzing many different three, four and five valve compensating and noncompensating instruments. Of particular interest is a five valve compensating instrument having a variance of 2.37% and standard deviation of 2.48%. The fifth valve lowers the instrument by 6 semitones. Instead of the fifth valve aforementioned a fifth valve of 0.237 of the length of the fourth valve could be used in combination with the fourth valve making the horn lighter. An Excel spread sheet is presented for the calculations relating to many ordinary and compensating three, four and five valve instruments.



### *Contributed Papers*

**3:15**

**2pMUb2. Musical acoustics education in Indonesia.** Gea O. Parikesit (Universitas Gadjah Mada, Jalan Grafika 2, Yogyakarta 55281, Indonesia, geaofp@yahoo.com) and Indraswari Kusumaningtyas (Dept. of Mech. and Industrial Eng., Universitas Gadjah Mada, Yogyakarta, Indonesia)

As the world's largest archipelago, Indonesia has a vast natural and cultural diversity, and hence a huge variety of musical instruments. Musical acoustics is, therefore, an important field of research and study in Indonesia. However, this subject is still rarely touched in undergraduate education. To help improve the quality of musical acoustics education in Indonesia, we have developed several methods. First, we design a series of final projects for undergraduate engineering students, where they can implement various methods and techniques previously studied in class (e.g., system identification, experimental methods, computational analysis, engineering design) into actual practice. These projects help the students to gain a more comprehensive perspective on the various applications of their knowledge. Second, we form inter-disciplinary collaborations, where the students can learn with musicians, craftsmen, scholars, and conservators. Such collaborations help the students to understand that, by studying the musical instruments, they are supporting the conservation of their own heritage. We have been applying these methods with our students in Universitas Gadjah Mada, particularly in our research on the bundengan, a very unique bamboo-based musical instrument. Our experience indicates that these approaches can be effective to help improve the quality of musical acoustics education in Indonesia.

**3:35**

**2pMUb3. Design, construction, and analysis of ukuleles as a method for teaching on the acoustic properties of stringed instruments.** Nicholas Gangemi (Physical Acoust., U.S. Naval Res. Lab., 4555 Overlook Ave SW, Washington, DC 20375, nicholas.gangemi@nrl.navy.mil), Nicholas Dutz (Elec. Eng., The Catholic Univ. of America, Washington, DC), Amelia Vignola (Acoust. Signal Processing and Systems, U.S. Naval Res. Lab., Washington, DC), Sarthak Regmi, Diego Turo, and Joseph Vignola (Mech. Eng., The Catholic Univ. of America, Washington, DC)

This presentation will discuss an undergraduate course taught at Catholic University where the primary learning objective was for the students to understand how the design characteristics of a stringed instrument effects its acoustic properties. Previous student-led design projects have researched the acoustic properties of stringed instruments, notably guitars and five-string banjos, but no formal course has ever been offered on the subject. Further, all previous investigations utilized commercially available instruments as the test subjects, and thus, were limited in the design characteristics available to test. The course combined hands-on education about the construction of ukuleles, with subsequent measurement and analysis of the acoustic properties. Ukuleles were chosen as the test subject due to their simplicity compared to other stringed instruments, which allowed students with various levels of shop experience to participate in relevant aspects of lutherie. The ukuleles featured both conventional and non-conventional design characteristics, and were constructed with both traditional and non-traditional methods. Upon completion of the course, the students constructed nine different ukuleles, collected a large set of acoustic data, and completed preliminary analyses of the various instruments. Interest in the subject has persisted beyond the course, and several research projects on the subject have resulted.

# Session 2pNSa

## Noise: Forty-One Years of Responding to External Stimuli: A Session in Honor of Elliott Berger III

Cameron J. Fackler, Cochair
*3M Personal Safety Division, 7911 Zionsville Rd., Indianapolis, IN 46268*

Lauraine L. Wells, Cochair
*3M, 817 W. 4th St., Loveland, CO 80537*

William J. Murphy, Cochair
*Division of Field Studies and Engineering, Noise and Bioacoustics Team, National Institute for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998*

**Chair's Introduction—1:05**

### *Invited Papers*

**1:10**

**2pNSa1. Reconciling laboratory and real-world hearing protector testing: The method A versus method B debate.** Dan Gauger (Res., Bose Corp., Bose Corp, MS.15B, 100 Mountain Rd., Framingham, MA 01701, dan_gauger@bose.com)

A major area of exploration during Elliott Berger's tenure as chair of ANSI S12 working group 11 was the effort to harmonize laboratory testing of hearing protectors with typical performance in real-world use. S12.6-1997 introduced two variants of the Real-Ear Attenuation (REAT) method with different requirements for subject selection and guidance—Method A (experimenter-supervised fit) and Method B (subject fit). Under Elliott's leadership the working group also produced three papers documenting the motivation for these methods. This paper will summarize these papers then focus on what followed: work spanning 2003 to 2010 that explored the strengths and weaknesses of the two methods, as well as how to distill REAT data to ratings that best convey to an industrial hygienist, in a simple fashion, how much performance to expect from a hearing protector. This work, done in an effort to guide potential improvements to the Noise Reduction Rating by the Environmental Protection Agency, led to three new standards: S12.6-2008, S12.68-2009, and S12.42-2010 as well as a comprehensive paper the author produced with Elliott. This work exemplifies Elliott's incessant and, at times, exhaustively thorough quest for clarity and correctness in the standards process.

**1:30**

**2pNSa2. The journey to hearing protection fit-testing: from a doctoral project, to meeting a mentor, to a world recognized best practice.** Jeremie Voix (École de Technologie Supérieure, Université du Québec, 1100 Notre-Dame Ouest, Montréal, QC H3C 1K3, Canada, jeremie.voix@etsmtl.ca)

Introduced in the mid-1970s, the concept of hearing protection device (HPD) fit-testing had been used by several researchers to assess on an individual wearer the exact amount of attenuation provided by a given HPD. In 2000, however, as part of his doctoral work, the author developed an objective method, referred to as Field-MIRE, involving simultaneous measurement with two microphones and dedicated to a new type of instantly molded custom earplugs. The so-called SonoPass measurement system was commercialized by Sonomax and first introduced during the NHCA conference held in Arizona in 2001. This was also for the author the opportunity to first meet Elliott H. Berger, one of his most cited author and soon to become mentor. Their formal collaboration started in 2006 when the F-MIRE system was adapted to test non-custom HPD, became exclusive to AEARO company, and was globally introduced as E-A-Rfit Validation System. Then years of further collaborative work led to ANSI S12.71 standard adopted in 2018 for field attenuation estimation systems (FAES). Along the way, their joint writing for the Noise Manual handbook was remembered by one as "one of the most daunting training job" and by the other as "a truly fun and amazingly rigorous exercise."

1:50

**2pNSa3. Perspectives from the honoree: Serendipity, gratitude, and an exiguity of prognostication.** Elliott H. Berger (Berger Acoust. Consulting, 221 Olde Mill Cove, Indianapolis, IN 46260, eberger@compuserve.com)

As this is being written, I am excited, honored, and humbled but the mere notion that a session has been planned in my honor, even though the full implications of the title "…responding to external stimuli" still escapes me. I am presuming all will be revealed. The opportunity to close out the show, to have the last word as it were, was also a surprising and an unexpected gift. My goal will be to make it worth the audience's while. I will briefly review the accomplishments of which I am most proud, and then emphasize the importance of serendipity in our lives, especially mine, as when I stumbled across an obscure 1/6-page add in the back of *Physics Today* that led me to a career associated with the ground-breaking yellow foam earplug. The gratitude I have for fabulous mentors in my life such as Larry Royster, Ross Gardner Jr., Don Gasaway, and Mead Killion, will be expressed. Finally, I may share a few brief observations about what might come next in the fields of hearing protection and hearing conservation.

TUESDAY AFTERNOON, 8 DECEMBER 2020 — 2:50 P.M. TO 4:20 P.M.



## Session 2pNSb

## Noise, Psychological and Physiological Acoustics, and Structural Acoustics and Vibration: Perception of Vehicle Noise I

Patricia Davies, Cochair
*Ray W. Herrick Laboratories, Purdue University, 177 S Russell Street, West Lafayette, IN 47907-2099*

Daniel Carr, Cochair
*Purdue University, 177 S. Russell Street, West Lafayette, IN 47907*

**Chair's Introduction—2:50**

### *Invited Papers*

2:55

**2pNSb1. Soundscape: How do we perceive vehicle noise under the exceptional circumstances due the pandemic? Impact of COVID-19 upon the understanding of our acoustic environments.** Brigitte Schulte-Fortkamp (HEAD Genuit Foundation, Ebert Straße 30 a, Herzogenrath 52134, Germany, bschulte_f@web.de)

New findings about the possible decrease in experienced daily loudness are vastly less important than recognition about the meaning of the sounds in our new sonic reality. The recent pandemic has changed daily life in many respects. These changes have caused significant changes in soundscapes also in regard to its acoustic parameters and as well in their perception. The application of the Soundscape standard ISO 12913 1-3 will guide the understanding of the recent developments concerning the new living situations. This paper will discuss the ISO standard and the potentials of its use in regard to electric vehicles in these exceptional circumstances.

3:15

**2pNSb2. Annoyance at the point of emission versus immission for impulsive noises at train passings.** Christine Huth (Möhler + Partner Ingenieure AG, Prinzstr. 49, Augsburg 86153, Germany, christine.huth@mopa.de), Melissa Forstreuter, Robert Arlt, and Manfred Liepert (Möhler + Partner Ingenieure AG, Augsburg, Germany)

Impulsive components of the noise of passing vehicles can cause increased annoyance for the nearby inhabitants. For example, the periodic excitation of a damaged wheel creates a noise with a clearly noticeable fluctuation strength. This leads—in addition to the sound pressure level, which is also raised by the impulsive component—to a significant annoyance of the residents. Usually, such signals are recorded in the vicinity of the point of emission (in the distance of 7.5 m). The scope of the present study was to investigate the question to what extent the annoyance of the residents at the point of immission can be predicted by using these recorded signals. For this purpose, by means of subjective evaluations the annoyance of original signals at the point of immission was compared with that of recordings at the point of emission reduced in accordance with the distance law.

## *Contributed Papers*

**3:35**

**2pNSb3. Use of Type II sequence diffusers for the purpose of improving noise barrier effectiveness.** Prachee Priyadarshinee (Mech. Eng., National Univ. of Singapore, #21-256B, 38 College Ave. East, Singapore 138601, Singapore, a0135595@u.nus.edu), Kian-Meng Lim, and Heow Pueh Lee (Mech. Eng., National Univ. of Singapore, Singapore, Singapore)

The Type II Luke sequence diffusers have been known to redirect sound energy and act as a beam steerer at certain frequencies. If the incident sound from the top-edge of a noise barrier can be redirected away from the shadow zone by installing such a device, the insertion loss of the noise barrier can be significantly improved. Numerical analysis revealed that the use of multiple periods of Luke sequence diffusers as reflection phase gratings has potential to improve nose barrier effectiveness even more than using a single diffuser on the top-edge. Compared to a rigid T-barrier, improvement in overall A-weighted insertion loss of 8dBA was achieved by installing sound diffusers designed on the Type II luke sequence on the top-edge of the noise barrier.

**3:55**

**2pNSb4. Creating an acoustic simulation of noise pollution.** Rian Stephens (CSIS, Univ. of Limerick, Limerick V94 T9PX, Ireland, stephensrian@gmail.com)

This project utilises acoustic simulations to represent traffic noise pollution and prevention in affected residential areas. Noise has "wide-ranging adverse health, social and economic effects" (Goines, 2007, p. 287). Noise pollution 'is more severe and widespread than ever before' and is predicted to worsen with growth in highway traffic (Goines, 2007, p. 287). This paper presents the development of a three-dimensional acoustic model of traffic noise pollution. This three-dimensional acoustic simulation is a computer program that enables the user to walk through a modelled housing estate. Road traffic is programmed to drive along a motorway that runs adjacent to the estate. The user experiences traffic noise pollution in real-time while being given the tools to reduce it. The author examines the techniques of propagating noise pollution in three-dimensional space and how noise level is dependent on average speed, category of vehicle and the average number of vehicles passing through a road section (Kephalopoulos, 2014). Reflection and absorption techniques allow noise to move realistically in 3-D space (Funkhouser, 1998; Shearer, 2007; Murphy, 2007; Rober, 2007).

TUESDAY AFTERNOON, 8 DECEMBER 2020 — 2:50 P.M. TO 4:20 P.M.

# Session 2pNSc

## Noise, Architectural Acoustics, and Structural Acoustics and Vibration: Impact of Transportation Noise on Buildings

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

Benjamin E. Markham, Cochair
*Acentech, 33 Moulton Street, Cambridge, MA 02138*

**Chair's Introduction—2:50**

## *Invited Papers*

**2:55**

**2pNSc1. Auralizing the impact of transportation noise sources.** Kelsey Rogers (Acentech, 33 Moulton St., Cambridge, MA 02138, krogers@acentech.com) and Benjamin E. Markham (Acentech, Cambridge, MA)

Auralization is a powerful and cost-effective tool for communicating the impact of transportation noise sources on building occupants, and aiding the project team in making well-informed design decisions. An appropriately calibrated listening demonstration gives the owner, architect, and consultants the opportunity to directly experience annoyance, disturbance, and difficulty (or ease) of communication with others. This presentation will review lessons learned from practice to achieve an immersive and realistic listening experience, including suitable loudspeakers, recordings, receptor positions, transmission loss data, simulation and audio reproduction of moving sound sources. It will also cover important limitations of auralization in the context of transportation noise sources, including the difficulty of low frequency source characterization and transmission loss prediction, and the relevance of feelable vibration. Recent case studies of aircraft, rail, and automobile noise will be used to highlight the unique role of auralization in the design process of residential and commercial buildings.

**3:15**

**2pNSc2. An isolation case study of a vertical caisson wall directly adjacent to an underground rail vibration source.** Wilson Byrick (Pliteq, Inc., 131 Royal Group Cresent, Woodbridge, ON L4H 1X9, Canada, wbyrick@pliteq.com)

A large concrete tower planned for construction was proposed at a site immediately adjacent to underground rail. The underground rail passenger trains pass frequently at a rate of once per 2-4 min 18–20 h per day. Excavation began and a caisson wall was constructed facing the underground vibration source. Preliminary measurements were made by placing an accelerometer on the East facing caisson wall. Attenuation was required in the 20-63 Hz frequency range. An isolation layer (GenieMat DM) was added to the caisson wall. The presentation will illustrate how this was done in combination with water proofing of the foundation and other onsite complexities. Large steel tie-backs are used to hold the caisson wall in place and these were treated to prevent any direct path for vibration. Upon completion of construction, measurements are taken and the level differences are presented. Attenuation exceeded the design and building code criteria at all one-third octave bands of concern.

**3:35**

**2pNSc3. Calculation of base vibration isolation effectiveness of a tall building column using four-pole parameters and finite element analysis.** James T. Nelson (Wilson Ihrig, Emeryville, CA) and James E. Phillips (Wilson Ihrig, 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, jphillips@wilsonihrig.com)

An approach often proposed for mitigating vibration and ground borne/structure borne noise within a large building involves the installation of intervening vibration isolating elastomer springs beneath building columns at the foundation level (or higher). The input mechanical impedance (the ratio of input force to input velocity response) at the base of an isolated column and the source mechanical impedance of the structure supporting the isolation must be significantly greater than the mechanical impedance of the isolation spring. Resonances within the column and connected floor structures can significantly reduce the input impedance of the column at the resonance frequencies. The large, concentrated loads at the bases of columns generally dictate that the strain of the spring (elastomer, for example) be low to avoid degradation and excessive creep of the spring over the lifetime of the building, thus driving the size and stiffness. This paper presents a method for calculating the column resonances using simple four-pole parameter systems with comparison to results using a Finite Element Analysis (FEA) approach.

**3:55**

**2pNSc4. Statistical assessment of traffic noise patterns and COVID-19 effects.** Wayland Dong (Veneklasen Assoc., 1711 Sixteenth St., Santa Monica, California, CA 90404, wdong@veneklasen.com), John Lo Verde, and Samantha Rawlings (Veneklasen Assoc., Santa Monica, CA)

Road traffic noise assessments are based on noise indicators, including, at a minimum, average noise indicators like equivalent sound level (Leq) and single-event noise indicators like maximum sound level (Lmax). In the absence of guidance from regulatory agencies as to how these indicators should be defined and measured, previous work has included proposed definitions and methods for estimating the noise indicators and determining the uncertainty and reliability of the estimates. The analyses were based on statistical analyses of time history noise levels recorded at various cities across Connecticut and California as well as Dublin, Ireland [King *et al.*, Noise-con (2016)]. Previous work suggested that the observed patterns and methods were broadly applicable across a wide variety of road types and cities, based on measurements performed at different locations. The dramatic changes in traffic due to the COVID-19 pandemic allowed analysis of the effect of changing traffic volumes and diurnal patterns on the time history levels at given locations. This paper describes the effect of the pandemic on traffic patterns and what this demonstrates about the variability and robustness of the statistical estimation methods.

## Session 2pPAa

## Physical Acoustics, Musical Acoustics, and Biomedical Acoustics: Acoustical Measurements Through Optical Principles III

Gregory W. Lyons, Cochair
*Information Technology Laboratory, U.S. Army Engineer Research and Development Center, 3909 Halls Ferry Road, Vicksburg, MS 39180*

Thomas R. Moore, Cochair
*Department of Physics, Rollins College, Department of Physics, Box 2743, Rollins College, Winter Park, FL 32789*

**Chair's Introduction—1:05**

### *Invited Papers*

**1:10**

**2pPAa1. Interferometric measurements of supersonic projectile acoustic signatures.** Matthew G. Blevins (U.S. Army Engineer Res. and Development Ctr., Construction Eng. Res. Lab., 2902 Newmark Dr., Champaign, IL 61822, matthew.g.blevins@usace.army.mil), Gregory W. Lyons (U.S. Army Engineer Res. and Development Ctr., Information Technol. Lab., Vicksburg, MS), Carl R. Hart (U.S. Army Engineer Res. and Development Ctr., Cold Regions Res. and Eng. Lab., Hanover, NH), and Michael J. White (U.S. Army Engineer Res. and Development Ctr., Construction Eng. Res. Lab., Champaign, IL)

Supersonic projectiles cause disturbances in the air that propagate and coalesce to form acoustic signatures. In the far-field, these signatures approach ideal N-waves. However, near the projectile these signatures are closely related to the projectile's shape. To study the generation and formation of acoustic signatures from supersonic projectiles, an experiment was conducted using optical methods. A z-type schlieren imaging system and laser interferometer were used in conjunction with acoustic sensors to measure the pressure field surrounding supersonic projectiles of various sizes and shapes. The rate of change of the phase difference recorded by the interferometer is inverted to obtain the ballistic pressure field by assuming cylindrical symmetry and using an inverse Abel transform. The accuracy of these reconstructed time series is evaluated by comparing N-wave parameters, such as period and peak pressure, with analytical results derived from the Whitham F-function. Improved rise time resolution obtained from the interferometric measurement is demonstrated by comparison with recordings from standard condenser microphones. Advantages of measuring acoustic signatures near the projectile by interferometric methods are discussed.

**1:30**

**2pPAa2. Free-field schlieren measurements of laser-induced breakdown shock propagation.** Carl R. Hart (U.S. Army Engineer Res. and Development Ctr., Cold Regions Res. and Eng. Lab., 72 Lyme Rd., Hanover, NH 03755, carl.r.hart@erdc.dren.mil), Gregory W. Lyons (U.S. Army Engineer Res. and Development Ctr., Information Technol. Lab., Vicksburg, MS), and Michael B. Muhlestein (U.S. Army Engineer Res. and Development Ctr., Cold Regions Res. and Eng. Lab., Hanover, NH)

Measurements of free-field shock propagation are fundamental to understanding source characteristics, and for transfer functions through complex environments. Probe interference and undesirable high-frequency response plague typical approaches with acoustic microphones, which are also limited to resolving the pressure field at a single position. Measurements made with optical methods do not have such drawbacks, and schlieren measurements are particularly well suited to measuring both the spatial and temporal evolution of nonlinear pulse propagation. A laser induced breakdown source is used to generate spherically symmetric weak shockwaves. A z-type schlieren instrument and high-speed framing camera are used to measure schlieren images of passing shock waves. Quantitative measurements relate schlieren image intensity values to light refraction angles, and by the inverse Abel transform the density (and pressure) fields induced by the shockwave are recovered. Comparisons of analytical predictions, assuming spherically symmetric and inviscid propagation, with experimentally measured shockwaves are presented.

### *Contributed Papers*

1:50

**2pPAa3. Acoustic measurements of capillary-gravity water surface waves.** Robert L. Lirette (Dept. of Phys. and Astronomy, The Univ. of MS, 502 Park Ln., Oxford, MS 38655, rlirette@go.olemiss.edu), Zheguang Zou (Dept. of Phys. and Astronomy, The Univ. of MS, University, MS), Guoqin Liu (Dept. of Phys. and Astronomy, The Univ. of MS, Oxford, MS), Xinyue Gong (Dept. of Phys. and Astronomy, The Univ. of MS, University, MS), and Likun Zhang (Dept. of Phys. and Astronomy, The Univ. of MS, Oxford, MS)

For this work an optical method for measuring surface acoustic waves was adapted to acoustics using airborne ultrasound for measurements on water surface waves. With ultrasound incident on periodic traveling water surface waves, the reflected signal can be treated as the diffraction pattern from a moving corrugated reflection grating as long as the amplitude of the water surface waves is much less than the incident acoustic wavelength. The acoustic signal received at the first-order diffraction maxima is amplitude modulated at the frequency of the water surface wave. The intensity of this modulation is directly proportional to the amplitude of the water surface wave. Using spectral decomposition of the signal, the water surface wave amplitudes are precisely determined at sum and difference frequencies around the source peak. The transmission of water surface waves incident on a solid piercing boundary was measured using this method to understand capillary-gravity wave interactions with boundaries. [Work supported by NASA.]

2:10

**2pPAa4. Geometric-acoustics analysis of single-scattering of nonlinearly evolving waves by circular cylinders.** Michael B. Muhlestein (U.S. Army Engineer Res. and Development Ctr., Cold Regions Res. and Eng. Lab., 72 Lyme Rd., Hanover, NH 03755, Michael.B.Muhlestein@usace.army.mil) and Carl R. Hart (U.S. Army Engineer Res. and Development Ctr., Cold Regions Res. and Eng. Lab., Hanover, NH)

Geometric acoustics, or acoustic ray theory, is used to analyze the scattering of high-amplitude acoustic waves incident upon rigid circular cylinders. Theoretical predictions of the importance of incorporating nonlinearity in modeling and the nonlinear evolution of the scattered wave field are provided. An analysis of scattering by many cylinders with a dilute concentration is also provided. The nonlinear evolution of the incident wave is shown to be of much greater importance to the overall evolution than the nonlinear evolution of the individual scattered waves. This presentation is based on an article recently published by the *Journal of the Acoustical Society of America*.



TUESDAY AFTERNOON, 8 DECEMBER 2020 2:50 P.M. TO 3:20 P.M.

## Session 2pPAb

## Physical Acoustics, Musical Acoustics, and Biomedical Acoustics: Acoustical Measurements Through Optical Principles IV

Gregory W. Lyons, Cochair
*Information Technology Laboratory, U.S. Army Engineer Research and Development Center, 3909 Halls Ferry Road, Vicksburg, MS 39180*

Thomas R. Moore, Cochair
*Department of Physics, Rollins College, Department of Physics, Box 2743, Rollins College, Winter Park, FL 32789*

**Chair's Introduction—2:50**

### *Contributed Paper*

2:55

**2pPAb1. Optically detected vibrations of crystal pendulum due to sun radiation.** Igor Ostrovskii (Phys. and Astronomy, Univ. of MS, 108 Lewis Hall Phys., University, MS 38677, iostrov@phy.olemiss.edu)

The optical experiments with suspended quartz crystals reflecting Laser-Doppler-Vibrometer (LDV) beam reveal an effect of crystal pendulum vibrations initiated by Sun's radiation of heavy particles. The LDV measures speed S(t) of oscillating crystal surface. The Fourier-transform of S(t) returns a spectrum of the acoustic vibrations with a few hertz frequency. The S(t) is a sine-type curve with amplitude depending on the time and space orientation of crystal axes. Maximum vibrational speed and corresponding crystal displacement from a position of equilibrium are observed when quartz piezoelectric axis is aimed toward the Sun. Experimental evidences exclude involvement of any electro-magnetic wave. The theoretical calculations for an oscillator driven by a periodic pulse-force are compared to the experimentally detected oscillations. The computations give a work done by an external force to move the crystal from its equilibrium to maximum displacement. It yields a mass of a thinkable particle that is necessary to initiate and support observed vibrations. In the case of individual particles propagating with a solar wind speed and hitting crystal, particle mass is of the order of 10 i×-21 kg. The physics of crystal-particle interaction may be related to gravitational pull of quartz atoms/ions and particle quantum properties such as matter waves.

## Session 2pPAc

# Physical Acoustics: General Topics: Bubbles II

Kevin M. Lee, Chair
*Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758-4423*

**Chair's Introduction—2:50**

### *Contributed Papers*

**2:55**

**2pPAc1. Weakly nonlinear acoustic theory on pressure propagation in bubbly flows with different cases of nonuniform distribution of initial flow velocities of gas and liquid phases.** Taiki Maeda (Eng. Mech. and Energy, Univ. of Tsukuba, Tsukuba, Japan), Tetsuya Kanagawa (Eng. Mech. and Energy, Univ. of Tsukuba, 1-1-1 Tennodai, Tsukuba, Ibaraki 305-8573, Japan, kanagawa.tetsuya.fu@u.tsukuba.ac.jp), and Takahiro Ayukai (Eng. Mech. and Energy, Univ. of Tsukuba, Tsukuba, Japan)

Three cases of weakly nonlinear propagation of plane progressive pressure waves in flowing water containing many uniformly separated spherical microbubbles are theoretically investigated, with a particular focus on different cases of initial flow patterns. The gas and liquid phases initially have a high velocity with uniform distribution, and a low velocity with nonuniform distribution for each phase. Basic conservation laws based on a two-fluid model are employed to govern different velocity distributions for gas and liquid phases. From the method of multiple scales with perturbation expansions, we derive three cases of nonlinear wave equation describing the long-range propagation of waves: (i) the KdVB (Korteweg–de Vries–Burgers) equation for a long wave, (ii) the NLS (nonlinear Schrödinger)-I equation for a short wave in slow nonuniform flow, (iii) the NLS-II equation for a short wave in fast nonuniform flow. As a result, all the initial velocities contribute to an advection effect of waves, initial nonuniform flow distribution induces a variable coefficient of advection term into the KdVB, NLS-I, and NLS-II equations, and an important role of relative velocity between the gas and liquid phases is clarified.

**3:15**

**2pPAc2. Weakly nonlinear theory and numerics on pressure wave propagation in a flowing water containing many translational bubbles acting a drag force.** Takahiro Yatabe, Takahiro Ayukai (Eng. Mech. and Energy, Univ. of Tsukuba, Tsukuba, Japan), and Tetsuya Kanagawa (Eng. Mech. and Energy, Univ. of Tsukuba, 1-1-1 Tennodai, Tsukuba, Ibaraki 305-8573, Japan, kanagawa.tetsuya.fu@u.tsukuba.ac.jp)

Focusing on the effect of translational motion of bubbles and drag force acting bubbles, we theotetically and numerically tackle weakly nonlinear propagation of plane progressive pressure waves in a compressible water flow uniformly containing a number of spherical gas bubbles. At an initial state, gas and liquid phases are flowing with each velocity. Drag force and virtual mass force are introduced in an interfacial transport across the bubble–liquid interface. As bubble dynamics, translation and spherically symmetric oscillation are considered. The bubbles do not coalesce, break up, disappear, and appear. The gas viscosity, the thermal conductivities of the gas and liquid are discarded. From the singular perturbation analysis up to the second order of approximation, we have two types of Korteweg–de Vries–Burgers (KdVB) equation with a newly introduced term without a differentiation due to the drag force from the basic equations for bubbly flows in a two-fluid model; the drag term is classified into the linear and nonlinear terms. The translation affects the nonlinear effect of waves and the drag force affects the dissipation effect of waves. We finally numerically solve the KdVB equation to compare three dissipation factors, i.e., acoustic radiation damping due to the liquid compressibility, liquid viscosity, and drag force for various initial void fraction and bubble radius.

**3:35**

**2pPAc3. Theoretical Improvement of a KZK equation for focused ultrasound in bubbly liquids with thermal effects.** Shunsuke Kagami (College of Eng. Systems, Univ. of Tsukuba, 1-1-1 Tennodai, Tsukuba, Ibaraki 305-8573, Japan, kagami.shunsuke.sw@alumni.tsukuba.ac.jp), Tetsuya Kanagawa, and Takahiro Ayukai (Eng. Mech. and Energy, Univ. of Tsukuba, Tsukuba, Japan)

Toward medical applications such as HIFU treatment, weakly nonlinear evolution of focused ultrasound in initially quiescent bubbly liquids is theoretically investigated with a special attention to the consideration of a thermal effect. Although our group (Kanagawa, *JASA*, 2015) was derived a Khokhlov-Zabolotskaya-Kuznetsov (KZK) equation for quasi-plane progressive ultrasound with initially nonuniform number density of microbubbles, thermal effects such as thermal conduction were completely neglected. In this paper, we modify our previous KZK equation by incorporating thermal effects, i.e., the thermal conduction and thermodynamics inside bubble, and the liquid viscous damping in bulk liquid. By using the method of multiple scales to basic equations for bubbly liquids introducing an energy equation proposed by Prosperetti (*JFM*, 1991), a revised KZK equation incorporating all the thermal effects was derived. Although a newly discovered dissipation term without differentiation corresponds to the thermal conduction, the coefficient of original dissipation term was expressed by the liquid viscosity and liquid compressibility (i.e., acoustic radiation damping). Hence, the physics of dissipation by thermal conduction differs that by liquid viscosity and compressibility. Furthermore, we emphasize that a thermal effect contributes the nonlinear effect, that is, nonlinear coefficient include the ratio of specific heats of the gas inside the bubble.

**3:55**

**2pPAc4. Derivation of an effective equation for ultrasound propagation in a liquid containing many encapsulated bubbles.** Yusei Kikuchi (Eng. Mech. and Energy, Univ. of Tsukuba, 1-1-1 Tennodai, Tsukuba, Ibaraki 305-8573, Japan, kikuchi.yusei.sp@alumni.tsukuba.ac.jp) and Tetsuya Kanagawa (Eng. Mech. and Energy, Univ. of Tsukuba, Tsukuba, Ibaraki, Japan)

Toward an establishment of basis for medical applications such as ultrasound diagnosis, the mathematical model for the propagation of ultrasound (or pressure waves) in a liquid containing microbubbles encapsulated by a viscoelastic shell is constructed from the theoretical viewpoint. For simplicity, the bubbles do not coalesce, break up, appear, and disappear; the bubbles are spherical, and these oscillations are spherically symmetric; the viscosity of gas inside the bubbles and the thermal conductivities of both phases are neglected: these assumptions are the same as those in our

previous studies [e.g., Kanagawa *et al.*, *J. Fluid Sci. Technol.* (2010); Kanagawa, *J. Acoust. Soc. Am.* (2015)]. In this paper, the Hoff model (equation of spherical shell bubble) is used for the equation of motion of the bubble in order to theoretically clarify the effect of viscosity and stiffness of the shell, instead of the Rayleigh-Plesset or Keller model. From the method of multiple scales, some linear and /or nonlinear wave equations (e.g., KdV-Burgers equation) as an effective equation are derived from the basic set of governing equations for liquids containing many encapsulated bubbles.

TUESDAY AFTERNOON, 8 DECEMBER 2020 1:05 P.M. TO 2:35 P.M.

## Session 2pPPa

## Psychological and Physiological Acoustics: Honoring William Yost's Contributions to Psychological Acoustics III

Robert A. Lutfi, Cochair
*Univ. of South Florida, Tampa, 4202 E. Fowler Avenue, Tampa, FL 33620*

Christopher A. Brown, Cochair
*Communication Science and Disorders, University of Pittsburgh, 4028 Forbes Tower, Pittsburgh, PA 15260*



**Chair's Introduction—1:05**

### *Invited Paper*

**1:10**

**2pPPa1. The WAY to lead—Reflections on the contributions of Dr. William A. Yost to the Speech and Hearing Science Program at ASU.** Yi Zhou (Speech and Hearing Sci., Arizona State Univ., 975 S. Myrtle Ave., Coor 3470, Tempe, AZ 85287, yizhou@asu.edu)

In the spring of 2007, after 30 years of Dr. Yost's leadership at the Parmly Hearing Institute at Loyola University Chicago, Dr. Sid Bacon invited him to join the Department of Speech and Hearing Science (SHS) at Arizona State University. Dr. Yost served as the department chair between 2007 and 2013. Dr. Yost brought a clear vision for how auditory research could grow to best position the department in the modern era of brain research. As department chair, he developed a strategic hiring plan to grow the scope of SHS research and expand faculty expertise in critical areas outside the traditional realm of SHS. This hiring strategy resulted in my coming to ASU. This talk will reflect on Dr. Yost's leadership of the department as well as his support for understanding central auditory mechanisms in my laboratory and the example he set for being a model scientist, a devoted teacher, and a productive scholar.

### *Contributed Papers*

**1:30**

**2pPPa2. Processing of tonal sequences and assessing user perception of audio products.** Sandra J. Guzman (Shure, Inc, 4800 W Touhy Ave., Niles, IL 60714, guzman_sandra@shure.com)

This talk will highlight how lessons learned from Bill Yost during my time at the Parmly Hearing Institute influenced both my past academic research and more recent efforts in commercial product development. Two primary subject areas will be points of focus, (1) discussion of my psychoacoustic research on the processing of tonal sequences, and (2) application of academic research methods to business concerns. The psychoacoustic study of tonal sequences evaluated involvement of musical training and working memory in the processing of tonal sequences defined by both pitch contour and sequence rhythm. To evaluate involvement of musical training in task performance, subjects were recruited from both music and audio-arts academic programs. Subjects reconstructed four-element tonal sequences by focusing on pitch only, rhythm only, or on both pitch and rhythm. Findings suggested that rhythm reconstruction was generally more difficult than pitch reconstruction, with effect of musical training primarily on processing sequences defined by pitch and effect of working memory on rhythmic sequences. Additionally, an overview of product development research will be presented, focusing on the assessment of end-user perception and understanding of audio products. The process, core challenges, and strengths and limitations of product development research will be examined.

**1:50**

**2pPPa3. A revisitation of Bill Yost's revisitation of how internal noise affects binaural detection.** Leslie R. Bernstein (Neurosci. and Surgery, Univ. of Connecticut Health Ctr., MC3401, University of Connecticut Health Ctr., Farmington, CT 06032, lbernstein@uchc.edu) and Constantine Trahiotis (Neurosci. and Surgery, Univ. of Connecticut Health Ctr., Farmington, CT)

Bill Yost's contributions to knowledge concerning auditory processing comprise several areas of study including binaural masking and discrimination, localizations of sounds (with both static and rotating listeners), binaural and monaural versions of repetition pitch, and mathematical approaches to understanding of each. Perhaps less well-known is his historical and analytical paper concerning estimation and measurement of "internal noise." In our view, that paper is also a fundamental contribution. Here, we report our related efforts to characterize and to measure types of internal noise that, on the one hand, limit monaural and/or binaural information processing and, on the other hand, help to explain why some listeners with slight, but clinically negligible, elevations in audiometric thresholds exhibit reliable and meaningful deficits in both binaural detection and binaural discrimination tasks. [Work supported by Office of Naval Research (N00014-15-1-2140; N00014-18-1-2473)]

**2:10**

**2pPPa4. The Yost approach to pitch perception: What we can learn from noise-like stimuli.** William P. Shofner (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, wshofner@indiana.edu)

One of Bill's major contributions to psychoacoustics is his innovative work using rippled noise (RN) to understand pitch. Bill has argued that RN-pitch percepts can be accounted for by models based on autocorrelation. Working with Bill, we showed that behavioral responses obtained from chinchillas to RNs generally paralleled those obtained from humans, suggesting the underlying processing is similar between the two species. Recent work from my lab using noise-vocoded harmonic tone complexes (NV-HTCs) was inspired by Bill's approach with RNs. NV-HTCs can also evoke pitch percepts like RNs, but present a challenge to Bill's autocorrelation approach, because they can have strong harmonic structures with weak or no periodicities. However, when NV-HTCs are passed through a gammatone filterbank model, weak stimulus periodicities become augmented in summary correlograms. An analytical model based on summary correlograms for NV-HTC responses from (1) humans in a magnitude estimation task and (2) chinchillas in a stimulus generalization task suggests that the underlying processing does indeed differ between the two species. Specifically, chinchillas appear to process the envelope whereas humans appear to process the fine structure. This talk will explore why this apparent processing difference exists, despite the similarity in chinchilla and human auditory filter bandwidths.

TUESDAY AFTERNOON, 8 DECEMBER 2020 — 2:50 P.M. TO 4:00 P.M.

## Session 2pPPb

## Psychological and Physiological Acoustics: Honoring William Yost's Contributions to Psychological Acoustics IV

Robert A. Lutfi, Cochair
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Christopher A. Brown, Cochair
*Communication Science and Disorders, University of Pittsburgh, 4028 Forbes Tower, Pittsburgh, PA 15260*

**Chair's Introduction—2:50**

### *Contributed Papers*

**2:55**

**2pPPb1. Where versus what: The relationship between speech understanding and auditory motion processing.** Michaela Warnecke (UW Madison, 151 E Wilson St., Apt. 505, Madison, WI 53703, elabelabohne@gmail.com) and Ruth Y. Litovsky (UW Madison, Madison, WI)

Most sounds involve motion, either because the sound source moves, or the listener does. Thus, perceiving moving sounds is critical for listeners' functional ability to navigate auditory environments. Studies to date focused on the limits of sound motion detection. By contrast, we are interested in the functional importance of auditory motion perception and aim to understand which acoustic cues govern listeners' classification of sound motion. In the present study, we hypothesized that auditory motion perception is influenced by the extent to which stimuli are perceived as intelligible. Utilizing Chimaera compositions of speech and spectrally matched noise tokens, we varied the envelope type and number of frequency channels of acoustic stimuli, manipulating speech intelligibility. Stimuli were presented in free field as stationary or moving. Normal-hearing adults judged sound motion in a 2-AFC task, and verbally repeated what they understood. Results show that a speech signal envelope biased sound motion judgments to be stationary, and increasing speech intelligibility further increased bias. This suggests that speech intelligibility influences auditory motion perception, and that sound motion and speech may not be processed simultaneously. [Work supported by NIH-NIDCD(R01DC8083), NIH-NICHD (U54HD090256) and ASA's Hunt Fellowship.]

3:15

**2pPPb2. Discrimination of interaural intensity differences and monaural intensity increments.** Mark A. Stellmack (Psych., Univ. of Minnesota, 75 East River Parkway, Minneapolis, MN 55455, stell006@umn.edu), Neal F. Viemeister, and Andrew J. Byrne (Psych., Univ. of Minnesota, Minneapolis, MN)

This study assessed the equivalence of a binaural task (two-interval discrimination of interaural intensity differences, IIDs) and a monaural two-interval increment-discrimination task, in which each interval consisted of two noise bursts of different intensities, with listeners identifying the interval with the larger intensity change. Stimuli were broadband noise, with trial-by-trial feedback provided. An overall level rove between intervals forced listeners to compare the two signals within each interval (across ears in the binaural task and across bursts in the monaural task). Both tasks involved presentation of four signals of different intensities, yielding the same optimal decision statistic. Assuming (1) discrimination of IIDs across intervals amounts to discrimination of lateral position and (2) variability in lateral position increases with IID leads to a prediction of Weber's law. Functions relating threshold IID delta (or monaural increment delta) in dB to the standard IID (or increment) in dB had nearly equal slopes in binaural (0.054) and monaural (0.058) conditions. While the derived binaural decision statistic is based on discrimination of an emergent percept of the stimulus (intracranial position), the similarity of results suggests that monaural increment-discrimination may be based on discrimination of a similarly emergent percept (a loudness step). [Work supported by NIH R01DC00683.]

3:35

**2pPPb3. Modeling the advantage of head-movements in judging elevation.** Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., School of Architecture, 110 8th St., Troy, NY 12180, braasj@rpi.edu), Jens Blauert (Bochum University, Inst. of Commun. Acoust., Bochum, NRW, Germany), M. Torben Pastore (College of Health Solutions, Arizona State Univ., Tempe, AZ), and Yi Zhou (Speech and Hearing Sci., Arizona State Univ., Tempe, AZ)

Estimating the elevation of sound sources is a challenging task for the auditory system due to the lack of interaural cues. In the 1930s, Wallach developed the first comprehensive theory on auditory elevation perception by means of auditory motion parallax during head movements. Years later, Blauert found that stationary listeners' perceived localization of narrowband sound sources correlated with the center frequencies of the narrowband sound sources rather than with their actual locations, leading to the idea of directional bands. Returning to Wallach's original idea, which has been modeled also by Zhong, Sun, and Yost in a robot hearing context, a new binaural algorithm will be presented to estimate elevation by auditory motion parallax. The model uses a front end with gammatone filterbanks and cross-correlators to compute binaural activity maps as a function of azimuth, elevation, and frequency. By continuously compensating for head rotations in the horizontal plane, all but the actual sound source position wash out, removing ambiguous positions from the cone of confusion. We will simulate binaural data in two primate species with vast differences in head size—humans and marmosets—to evaluate the generality of using the strategy of auditory motion parallax in elevation localization. [Work supported by NSF BCS-1539276, BCS-1539376, NIH F32 5F32DC017676 & ERC #618075.]

## Session 2pSAa

# Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics: Acoustic Metamaterials V

Christina J. Naify, Cochair
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Alexey Titovich, Cochair
*Naval Undersea Warfare Ctr., Carderock Division, West Bethesda, MD 20817-5700*

Bogdan-Ioan Popa, Cochair
*Univ. of Michigan, Ann Arbor, MI 48109*

**Chair's Introduction—1:05**

## *Contributed Papers*

**1:10**

**2pSAa1. Acoustic metasurfaces as sound diffusers and diffsorbers.** Janghoon Kang (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX), Samuel D. Parker (Mech. Eng., Univ. of Texas at Austin, Austin, TX), Joe Skeens (Aerosp. Eng. and Eng. Mech., The Univ. of Texas at Austin, Austin, TX), Benjamin C. Treweek (Computational Solid Mech. & Structural Dynam., Sandia National Labs., Albuquerque, NM), Timothy Walsh (Simulation Modeling Sci., Sandia National Labs., Albuquerque, NM), and Michael R. Haberman (Walker Dept. Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@utexas.edu)

Acoustic metasurfaces (AMS) may offer additional design space for the creation of sound diffusers and diffsorbers. We present the design and experimental characterization of an AMS created to diffuse sound fields using a treatment of uniform thickness. The AMS unit cell designs were chosen to minimize viscous losses while enabling control of reflected phase at each point of the AMS. The design geometry was then optimized with a parametric sweep of unit cell dimensions using finite element (FE) models where the target phase shifts are associated with a one- and two-dimensional quadratic residue diffuser (QRD). Samples were fabricated using additive manufacturing and impedance tube measurements were conducted to validate FE predictions. Modeled and measured reflection coefficients were shown to be in good agreement. The scattered field was then measured in an anechoic chamber using the logarithmic frequency modulated chirp excitation and the deconvolution methods to provide narrowband and broadband estimates of sound diffusion performance. Overall performance is presented for narrowband and broadband considerations. The AMS response is compared and contrasted with traditional QRD designs, highlighting differences in frequency- and angle-dependence as well as temporal diffusion and absorption. [S.N.L. is managed and operated by NTESS under DOE NNSA Contract No. DE-NA0003525.]

**1:30**

**2pSAa2. Characterizing the physical limitations of phase-only metasurface design for acoustic holograms.** Samuel D. Parker (Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX 78712, sdparker@utexas.edu), Janghoon Kang (Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX), Benjamin C. Treweek (Computational Simulation, Sandia National Labs., Albuquerque, NM), Joe Skeens (Aerosp. Eng. and Eng. Mech., Univ. of Texas at Austin, Austin, TX), Timothy Walsh (Computational Simulation, Sandia National Labs., Albuquerque, NM), and Michael R. Haberman (Walker Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Numerous recent studies have shown that acoustic metasurfaces (AMS) can be used to control reflected or transmitted wave fronts. Acoustic holograms have commonly been created by designing a metasurface that applies local phase changes to a wave front to construct arbitrary pressure fields at one or more image planes. The hologram design process typically features iterative methods, such as the Iterative Angular Spectrum approach. However, metasurfaces that apply phase-only modulation are restricted in the quality of the pressure field reconstruction that can be achieved, especially when attempting to create different holograms at different planes for a single-frequency incident wave. The control of magnitude and phase by an AMS is of significant interest because it enables one to completely define an entire reflected or transmitted wave field in a 3-D region. In the present work, the physical limitations in simultaneously reconstructing multiple single-frequency holograms at different planes via phase-only metasurface design are examined. Methods for overcoming these limitations, including both amplitude and phase control by an acoustic metasurface, are also investigated. Comparisons between simulation and experiment are made for a multi-image reflective hologram system. [S.N.L. is managed and operated by NTESS under DOE NNSA Contract No. DE-NA0003525.]

1:50

**2pSAa3. Reflective acoustic holograms for arbitrary pressure patterning and focusing.** Ahmed Sallam (Mech. Eng., Virginia Tech, 495 Old Turner St., Blacksburg, VA 24060, ahmedsallam@vt.edu), Vamsi C. Meesala (Biomedical Eng. and Mech., Virginia Tech, Blacksburg, VA), and Shima Shahab (Mech. Eng., Virginia Tech, Blacksburg, VA)

Monolithic passive holographic lenses (MPHLs) are being proposed as an alternative technology to phased array transducers (PATs) and subwavelength structured metamaterials due to their unprecedented spatial control over the sound field. These new capabilities enable precise focusing of acoustic energy or high resolution elaborate pressure patterning without the limitations associated with the aforementioned techniques. In particular, PATs and metamaterials require either an impractical high number of active elements or face fabrication limitations when operated at ultrasonic frequencies, especially underwater or inside the human body. MPHLs do not have such limitations and are being utilized in a wide range of crucial ultrasonic applications such as medical imaging and therapy, ultrasonic contactless energy transfer and nondestructive testing. In this work, we introduce a 3-D-printed metal reflective acoustic hologram. We will present the modeling and design process of such holograms as well as their capabilities for constructing single and multifocal spatial distributions of acoustic energy in a target plane. The effects of operating frequency, hologram material and propagation distance will also be highlighted. The results are verified using finite element multiphysics simulations in COMSOL as well as experimental investigations. [This work was supported by NSF Grant Nos. ECCS—1711139 and IIP-1738689, which are gratefully acknowledged.]

2:10

**2pSAa4. "Auxauralities" : Ears-on 3-D-printed acoustics metamaterials for sound art.** Georges Roussel (GAUS, Université de Sherbrooke, 2500 Blvd de l'Université, Sherbrooke, QC J1K 2R1, Canada, georges.roussel@usherbrooke.ca), Ana Dall'Ara-Majek (Université de Montréal, Montréal, QC, Canada), François Proulx (GAUS, Université de Sherbrooke, Sherbrooke, QC, Canada), Philippe-Aubert Gauthier (UQAM, Montréal, QC, Canada), and Nicolas Bernier (Université de Montréal, Montréal, QC, Canada)

Due to unique acoustic properties, acoustic metamaterials (AMMs) find many engineering applications in industrial or military sectors. However, as a result of their complex behavior and since they are generally unknown to the non-scientists, AMMs are not integrated into the thinking of daily auditory culture and sound environments. AMMs' potentially speculative effects on sound environments and audio cultures could be communicated and investigated through art. To fully seize this opportunity, there is a need for Arts and Sciences sympoiesis as a way of thinking, creating, designing and experientially testing audible-range AMMs. Since engineering AMMs require considerable expertise in physics and additive manufacturing, it is not easy for non-acousticians to explore such materials. Therefore, an "ears-on" audible experiential approach was developed. To do so, the presented work is a co-creation process for developing tools for sound art. Two approaches were investigated as sound art prototyping tools. First, an auralization process is presented to perform acoustic simulations and virtually hear the results beforehand. A second approach uses modular 3-D-printed resonators and crystals for the rapid improvement of the hands-on and "ears-on" iterative design of a sound art installation: "Auxauralities" [Work supported by Fonds Recherche Québec Audace, 2020-AUDC271071].

# Session 2pSAb

## Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics: Acoustic Metamaterials VI

Christina J. Naify, Cochair
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Bogdan-Ioan Popa, Cochair
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**Chair's Introduction—2:50**

### *Contributed Papers*

**2:55**

**2pSAb1. Elastic wave propagation in structures with triply periodic minimal surfaces.** Maria Carrillo-Munoz (Aerosp. Eng., Wichita State Univ., 1845 Fairmount St., Wichita, KS 67260, mxcarrillomunoz@shockers.wichita.edu), Nelson Caceres, Saranchana Keattitorn, and Bhisham Sharma (Aerosp. Eng., Wichita State Univ., Wichita, KS)

We study the elastic wave propagation behavior of triply periodic minimal surface structures. Triply periodic minimal surfaces (TPMS) are surfaces with periodicity in three independent directions and a mean surface curvature of zero. Here, we apply the finite element method to obtain the dispersion curves of six different TPMS structures: Schwarz' P, Schwarz' D, Gyroid, Fischer-Koch's S, Schoen's I-WP, and Schoen's F-RD. The dispersion curves are obtained by modeling their respective unit cells, appropriately applying the Floquet-Bloch periodic boundary condition, and then performing an eigenfrequency analysis while sweeping over the boundaries of their respective irreducible Brillouin zones. We then classify each band according to their respective propagation modes (*P*- or *S*-waves) and study the effect of surface topology on each mode. Finally, we discuss the generation of directional and polarized bandgaps in these structures.

**3:15**

**2pSAb2. Elastic wave propagation in self-similar and non-self-similar hierarchical two-dimensional square lattices.** Refugine Nirmal Ignacy Muthu, Maria Carrillo-Munoz (Aerosp. Eng., Wichita State Univ., Wichita, KS), and Bhisham Sharma (Aerosp. Eng., Wichita State Univ., 1845 Fairmount St., Wichita, KS 67260, bhisham.sharma@wichita.edu)

In this talk, we examine the elastic wave propagation behavior of infinite two-dimensional self-similar and non-self-similar hierarchical square periodic lattices. Using finite element analysis, we calculate the dispersion curves of different hierarchical lattice structures along their irreducible Brillouin zone (IBZ) paths. The effects of similarity, geometrical symmetry, and material symmetry on bandgap generation are studied. The mechanisms responsible for bandgap generation are explained by studying the mode shapes for different characteristic length ratios. We show that material symmetry can help induce wave attenuation bandgaps at low frequencies. Further, we classify the *P*- and *S*-wave propagation modes and study the effect of similarity on the generation of polarized bandgaps. We demonstrate that controlling the hierarchical similarity provides a robust method for tailoring the dispersion characteristics of periodic lattice structures.

**3:35**

**2pSAb3. Predicting wave propagation in periodic lattices with internal resonators.** Matthew Kelsten (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, mjk308@scarletmail.rutgers.edu) and Andrew N. Norris (Mech. and Aerosp. Eng., Rutgers Univ., Piscataway, NJ)

Periodic lattice metamaterials like that of the square honeycomb, hexagonal honeycomb, and kagomè lattices have proven themselves useful in structural applications with their ability to strongly suppress vibrations over a broad range of frequencies at subwavelength scales. There have been many attempts in recent years to supplement their dynamic features with the use of internal resonators, creating additional band gaps for heightened overall performance. However, the broad definition of an internal resonator in this context means that it can be difficult to model all cases extensively. In an attempt to help bridge the analytical gap, we consider 2-D lattice designs with incorporated cantilever beam-type internal resonators. Doing so allows for a FEM technique to be used that treats each lattice, and its embedded resonators, as a collection of Timoshenko beams. With it, we can create global mass and stiffness matrices that serve as the driving force behind the characterization of band diagrams and their modal structure. A semi-analytical modeling approach is introduced which highlights the explicit dependence of band diagram features on resonator parameters. The general methodology is discussed, and numerical results are compared with simulated results generated through COMSOL Multiphysics.

**3:55**

**2pSAb4. Three-dimensional periodic multifunctional lattice materials for simultaneous vibration isolation and heat conduction.** Ganesh U. Patil (Mech. Sci. and Eng., Univ. of Illinois Urbana-Champaign, 144 Sidney Lu Mech. Eng. Bldg., MC-244, 1206 West Green St., Urbana, IL 61801, gupatil2@illinois.edu), Oluseyi Babatola, Daniel Hsieh, Sanjiv Sinha, and Kathryn Matlack (Mech. Sci. and Eng., Univ. of Illinois Urbana-Champaign, Urbana, IL)

Thermomechanical applications need materials with higher thermal conductivity and vibration mitigation capability; these functionalities are often interconnected and difficult to optimize simultaneously. Lattice materials are known to exhibit bandgaps by virtue of their rational structural design and periodicity; however, they are inherently thermally resistive, limiting their use in thermomechanical applications. Past efforts towards designing multifunctional materials are focused either on static-dynamic or static-thermal properties. In this talk, we present a design platform for a three-dimensional multifunctional lattice material for simultaneous vibration mitigation and heat conduction within the same structure. We integrate "*heat pipes*" inside lattice trusses to improve the thermal conductance while we vary the material distribution at the truss junctions, denoted as "*nodal mass*," to generate full bandgaps. Through numerical analysis, we show independent control of both functionalities, showing the versatility of our design. Our design achieves an increase in specific thermal conductance by 3 orders-of-magnitude while opening full bandgaps at structural vibration frequencies around 2–10 kHz. Further, we study the anisotropic thermal conductance and the effect of nodal mass fractions on bandgaps of various lattices to facilitate their selection for a specific application. This study opens up the application of lattice materials for demanding environments.

TUESDAY AFTERNOON, 8 DECEMBER 2020 1:05 P.M. TO 2:35 P.M.

## Session 2pSCa

## Speech Communication and Psychological and Physiological Acoustics: Reintroducing the High-Frequency Region to Speech Perception Research III

Ewa Jacewicz, Cochair
*Speech and Hearing Science, The Ohio State University, 1070 Carmack Road, 110 Pressey Hall, Columbus, OH 43210*

Robert A. Fox, Cochair
*Speech and Hearing Science, The Ohio State University, 1070 Carmack Road, 110 Pressey Hall, Columbus, OH 43210-1002*



**Chair's Introduction—1:05**

### *Contributed Papers*

**1:10**

**2pSCa1. Fricative variability in normal adult speakers.** Christine H. Shadle (Haskins Labs., 300 George St. Ste. 900, New Haven, CT 06511, shadle@haskins.yale.edu), Wei-rong Chen (Haskins Labs., New Haven, CT), Laura Koenig (Haskins Labs., New Haven, NY), and Jonathan L. Preston (Communications Sci. & Disord., Syracuse Univ., Syracuse, NY)

This study aims to establish a model for the variability of fricatives in normal speakers. Such a model is useful for studying populations with disordered speech (e.g., laryngectomies, glossectomies, adolescents with residual speech sound errors, etc.). Seven normal adult speakers (4 women, 3 men) were recorded uttering a corpus containing isolated words and connected sentences, including all English fricatives in a variety of phonetic contexts. Multitaper spectra of the fricatives were computed. For each spectrum, parameters estimating the frequency of the main peak ($F_M$), amplitude difference between that peak and the low-frequency minimum (AmpD), and high-frequency energy (LevelD and spectral tilt) were computed. These have been shown to be closely related to, respectively: place of the constriction, strength and degree of localization of the noise source, and changes in the energy distribution of the source during the fricative. Comparison to other studies in which these parameters have been used can extend their findings to all English fricatives instead of only /s/ or only sibilants, and test the assumptions underlying the definitions of the parameters. In turn, the means and variabilities computed allow comparisons to pathological speech, and allow firmer conclusions about the causes of any deviation from normal parameter values.

**1:30**

**2pSCa2. Contributing spectral regions to subjective intelligibility of dysphonic speech in noise.** Keiko Ishikawa (Speech and Hearing Sci., Univ. of Illinois, Urbana-Champaign, 901 S. 6th St., Champaign, IL 61820, ishikak@illinois.edu) and Brian B. Monson (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Dysphonia negatively affects speakers' intelligibility in noisy places. Although this problem is well-recognized, little is known about the relationship between acoustics and intelligibility of dysphonic speech. The purpose of this study was to identify spectral regions critical to the intelligibility deficit in dysphonic speech. Sentences from the Consensus of Auditory-Perceptual Evaluation of Voice were recorded from 18 speakers with dysphonia and 3 age-gender matched speakers with healthy voice. The intensity of the speech samples was normalized at 60 dB SPL, and cafeteria noise was added to these recordings at a signal-to-noise ratio (SNR)-3, 0, and + 3. Perceived (i.e., subjective) intelligibility of these stimuli was rated on a 7-point scale by 45 native speakers of American English with normal hearing. Spearman rank correlation tests were conducted to evaluate the association between subjective intelligibility ratings and spectral energy in the f-ollowing frequency regions: 0–8 kHz, 8–16 kHz, 0–1 kHz, 1–2 kHz, 2–4 kHz, 4–6 kHz, 6–8 kHz, 8–10 kHz, 10–12 kHz, 12–14 kHz, and 14–16 kHz. The results indicated that the energies in 8–16 kHz, 0–1 kHz, 1–2 kHz, 6–8 kHz, and 10–12 kHz ranges were significantly associated with the subjective intelligibility ratings. The clinical relevance of the findings will be discussed.

**1:50**

**2pSCa3. High-frequency spectral shaping enhances glimpsing speech in noise for younger adults with normal hearing.** Rachel Madorskiy (Speech, Lang., Hearing and Occupational Sci., Univ. of Montana, 32 Campus Dr., Missoula, MT 59812, rachel.madorskiy@mso.umt.edu), Daniel Fogerty (Dept. of Commun. Sci. and Disord., Univ. of South Carolina, Columbia, SC), and Judy R. Dubno (Dept. of Otolaryngology-Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

Amplification is necessary to restore audibility to individuals with hearing impairment. However, frequency-specific amplification alters the spectrum of speech in addition to increasing the speech level. The current study investigated the effects of spectrum and speech level on speech recognition by younger adults with normal hearing. Sentence recognition was measured in quiet, speech-modulated noise, and steady-state noise. In addition, acoustic metrics were utilized to quantify the alterations resulting from spectral shaping. Acoustic metrics included (1) the Hearing Aid Speech Perception Index, in order to investigate changes to the temporal envelope, and (2) glimpse analyses, to quantify the proportion of time-frequency units that were glimpsed in a sentence. Results demonstrated that at high speech levels in quiet, spectral shaping was associated with lower sentence recognition. However, in all other conditions, and particularly in noise, spectral shaping enhanced sentence recognition. Acoustic metrics indicated that this enhancement in noise is associated with increases in the proportion and signal-to-noise ratio of glimpses, which resulted in improved preservation of the envelope and temporal fine structure of sentences in comparison to unshaped stimuli. Thus, spectral shaping is beneficial for speech recognition in noise for normal hearing participants due to enhanced speech glimpses.

**2:10**

**2pSCa4. The role of extended high frequencies in children's speech-in-speech recognition.** Mary M. Flaherty (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S Sixth St., Champaign, IL 61820, maryflah@illinois.edu), Kelsey Libert, and Brian B. Monson (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

This study examined the extent to which school-aged children (5–17 yrs) are able to take advantage of extended high frequency (EHF) energy when recognizing sentences in a two-talker masker. Recent work demonstrated that EHFs can be beneficial for adults' speech-in-speech recognition, but its role in children's speech recognition has not been studied. Given that children have superior EHF hearing compared to adults and that EHF audibility may play an important role in language development, we hypothesized that EHF may be especially useful to children in multitalker contexts. The present study measured children's open-set sentence recognition in a two-talker masker using two filtering conditions: full band versus all stimuli low-pass filtered at 8 kHz. Given that EHF energy emission in speech is highly dependent on the head orientation of the talker, two masker head rotation conditions were tested: both maskers at 45 deg or at 60 deg rotation, relative to the target talker. Preliminary results demonstrate children performed best when EHF was present, indicating they use EHF for speech-in-speech recognition. However, thresholds remained elevated compared to adults, suggesting that while EHF is a salient cue for children, their increased EHF hearing sensitivity (relative to adults) did not increase the EHF benefit.

TUESDAY AFTERNOON, 8 DECEMBER 2020 — 2:50 P.M. TO 3:35 P.M.

## Session 2pSCb

## Speech Communication: Speech Articulation I (Poster Session)

Authors will be at their posters from 2:50 p.m. to 3:35 p.m.

### *Contributed Papers*

**2pSCb1. X-ray analysis of velum movement in continuous speech.** Jahurul Islam (Linguist, Univ. of BC, 2613 West Mall, Totem Field Studios, Vancouver, BC, Canada, jahurul.islam@ubc.ca), Gillian de Boer, Chiachih Lo, Hillary Smith, Ernest Tse, and Bryan Gick (Linguist, Univ. of BC, Vancouver, BC, Canada)

In speech, the velopharyngeal port (VPP) is closed for oral sounds and open for nasal consonants and nasalized vowels. While VPP opening has been much studied, the stimuli have typically been sustained vowels or short CVC words. To study velum movement in continuous speech, we examined X-ray film database data [Munhall *et al.*, *J. Acoust Soc. Am.* **98**(2), 1222–1224 (1995)] from French and English speakers. The films' speech audio were annotated and velum movement in each frame was tracked using videokymography in ImageJ. Degree of VPP opening was normalized by speaker and plotted in arbitrary units (au). Each velum opening and closure excursion created a peak (max value 1.0) with a corresponding opening and closing slope. Preliminary analysis of two English and six French speakers shows the average VPP opening was greatest during inter-utterance pauses at 0.54 au (SD 0.24), while for nasals the opening was 0.34 (SD 0.17) and for oral sounds the velum was generally closed. There was greater mean opening for utterance-final nasals [0.41 SD(0.21)] than utterance-initial [0.25 SD(0.13)] or mid-utterance [0.32 SD(0.15)] nasals. Across-speaker slope analysis suggests the velum moves at a similar speed for opening and closing movements. Across-language comparison will be reported.

**2pSCb2. Internal gestural coordination for derived long nasals.** Miran Oh (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, miranoh@usc.edu) and Dani Byrd (Linguist, Univ. of Southern California, Los Angeles, CA)

Nasal consonants have been studied mainly by investigating acoustic nasal energy or the oral gesture's constriction, while less attention has been paid to how velic and oral components in nasals temporally interrelate. Given that nasals essentially require specific timing between oral and velum gestures—that is, oral closure must occur when the velum is sufficiently

low—it is important to understand the regularity of this coordination. By examining the relative timing of oral and velum gestures in nasals of Seoul Korean, this real-time MRI study aims to illuminate the gestural organization of various nasal structures. Target stimuli include onset nasals (/#n/), coda nasals (/n#t/ & /n#p/), and juncture geminate nasals (both concatenated /n#n/ and assimilated /t#n/). Findings show that concatenated and assimilated geminates crucially differ in their gestural coordination and their variability. Concatenated geminates and coda nasals show similar oral-velum timing relations, while some assimilated geminates show an onset-like coordination. Furthermore, while both geminates show more extended overlap between oral and velum gestures than onset nasals, assimilated geminates exhibit greater interspeaker variability. This suggests that the internal coordination for assimilated nasal geminates is less rigidly (or stably) organized, exhibiting characteristics of both singleton onset nasals and geminate nasals. [Work supported by NIH.]

**2pSCb3. The relationship between segment differentiation strategies and articulatory-acoustic variability across speakers.** Sarah Harper (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089-1693, skharper@usc.edu)

Although variability is prevalent in speech production, individual articulatory and acoustic dimensions differ in their propensity towards variation for any given segment (e.g., Hillenbrand *et al*., 1995; Whalen *et al*., 2018). Proposals tying these differences in variability to a dimension's involvement in phonological contrast (e.g., Keating, 1990) are complicated by the observation of interspeaker differences in both the amount of variability exhibited along certain dimensions (e.g., Harper & Goldstein, 2019) and these dimensions' use in segment differentiation (e.g., Schertz *et al*., 2015). This study examines whether individual differences in the amount of variability exhibited along specific articulatory and acoustic dimensions are related to interspeaker variation in the contribution of these dimensions to differentiating segment pairs. Articulatory and acoustic data from America English speakers in the Wisconsin X-Ray Microbeam Corpus (Westbury, 1994) were analyzed to characterize both the variability of specific dimensions in /s/, /ʃ/, /l/, and /ɹ/ and the contribution of each dimension to the production of the /s/∼/ʃ/ and /l/∼/ɹ/ contrasts for individual speakers. Preliminary results suggest co-variation on the level of the individual between the extent of variability observed along particular dimensions and dimensions' contribution to differentiating the examined segment pairs. [Work supported by NIH].

**2pSCb4. Articulatory and acoustic phonetics of voice actors.** Colette Feehan (Linguist, Indiana Univ., Bloomington, 107 s Indiana Ave., Bloomington, IN 47404, cmfeehan@indiana.edu)

Voice actors are an under-utilized population for linguistic study. This type of vocal performance requires actors to perform careful, consistent, and precise vocal tract manipulations to portray different character types such as imitating a dialect or performing complex manipulations that simulate a smaller vocal tract to sound younger. There has been relatively little research focusing on voice actors' unique vocal tract manipulations. Some previous studies have looked at how actors alter laryngeal setting, to portray specific characters types in animation (Teshigawara and Murano, 2004; Starr, 2015), but investigations of articulatory manipulations employed are rare. Even more rare are investigations focusing on these articulatory manipulations and how they link to the acoustic output. This study uses 3-D ultrasound and acoustic analyses to compare the kinds of articulatory strategies that actors use to approximate a smaller vocal tract and achieve a 'childlike' acoustic percept. Preliminary analyses indicate that actors have many different types and combinations of manipulations that they can implement including hyoid bone and laryngeal raising, gesture fronting, tongue grooving, lip movement, and F0 manipulation. Despite these differences in approach, the actors still achieve similar child-like percepts. This poster compares strategies from 3 professional and 3 amateur actors (4 male and 2 female) and will describe within-subject variation across each actor's adult and imitated child voices.

**2pSCb5. Laryngeal-acoustic relations in smiled speech.** Gillian de Boer (Linguist, Univ. of BC, Stores Rd. Annex, Stores Rd. Annex, Vancouver, BC V6T1Z4, Canada, gillian.deboer@ubc.ca), Donald Derrick (Univ. of Canterbury, Christchurch, New Zealand), Murray Schellenberg, and Bryan Gick (Linguist, Univ. of BC, Vancouver, BC, Canada)

Smiling is a social signal that can be both seen and heard. Smiling can increase speech amplitude and raise F0 and formants. However, experimental research on the role of larynx height in smiled speech is limited. 21 English speakers (6 M) repeated words in a carrier phrase with a neutral face or while smiling. The participants were recorded with audio, video and laryngeal ultrasound. F0, F1 and F2 were extracted for the duration of target vowels /i/, /u/ and /a/. Ultrasound images of laryngeal position were measured using Optical Flow. The laryngeal and acoustic data were analyzed in R with linear mixed models with smiling condition, timepoint-in-vowel, and gender as fixed effects. There was a significant effect of timepoint-in-vowel for larynx height (raising towards the end) and a smile-timepoint interaction effect (the larynx raised more at the end for smiling condition). Acoustically, smiling led to significantly higher F0 across vowels, and significantly higher F1 and F2 for /a/ but not /i/ or /u/. F2 timepoints were significant for all three vowels (F2 trajectories differed) across smile conditions. Results indicate smiling has a consistent effect on larynx height and variable effect on specific speech sounds.

**2pSCb6. Articulation of non-normative variants of L1 and L2 Spanish trill /r/.** Ji Young Kim (Spanish and Portuguese, UCLA, 5310 Rolfe Hall, Box 951532, Los Angeles, CA 90095, jiyoungkim@ucla.edu), Matthew Faytak (Linguist, Univ. of California, Los Angeles, Los Angeles, CA), Gemma Repiso Puigdelliura, and Erin Mauffray (Spanish and Portuguese, UCLA, Los Angeles, CA)

This study examines within-speaker variability in L1 and L2 Spanish trill production. The Spanish trill is canonically produced with two or more brief contacts between the tongue tip and the alveolar ridge, Although multiple tongue-palate contacts is considered the norm, some non-normative variants across Spanish dialects involve fewer contacts with frication (e.g., tap followed by fricative; sibilant fricative). L2 phonology research shows that English speakers are known to associate frication with /r/ and produce frication for /r/ themselves. However, it is unclear whether these variants are articulated with similar tongue posture to those of native speakers or whether differences are due to aerodynamic factors. Synchronized ultrasound and acoustic recordings of trill production were made for three Spanish native speakers and four English L2 learners of Spanish. Trill realizations will be classified as normative trill, variants involving frication, or other from the acoustic signal. Comparison of tongue contours before, during, and after trill initiation will be carried out to examine the nature of the articulation of non-normative variants.

**2pSCb7. Lateral articulation of the Spanish trill /r/.** Erin Mauffray (Spanish and Portuguese, UCLA, 5310 Rolfe Hall, Box 951532, Los Angeles, CA 90095-1532, emauffray@g.ucla.edu), Matthew Faytak (Linguist, UCLA, Los Angeles, CA), Ji Young Kim (Spanish and Portuguese, UCLA, Los Angeles, CA), and Gemma Repiso Puigdelliura (Spanish and Portuguese, UCLA, Los Angeles, CA)

The Spanish trill /r/ represents a rich area for language acquisition research because of its particular articulatory requirements and its tendency toward variation. Lateral bracing has been documented during production of the canonical apical trill. Some articulatory studies suggest alternate lateral articulations of the Spanish trill, with lateral bracing on one side and vibration on the other, which are not distinguishable from central trills from the acoustic signal [Rivera-Campos and Boyce, "Describing alternative articulations of the Spanish trill /r/ by ultrasound technology," *JASA* **133**(5)]. With the aim of replicating these findings, the present study uses ultrasound imaging in midsagittal and coronal planes to characterize tongue position and movement during different phonetic variants of /r/ produced by seven speakers (four L2 Spanish learners and three native Spanish speakers). The study will shed light on the frequency of the central and lateral strategies in native Spanish speakers and L2 learners, whether the presence of frication or failure of trilling is connected to the lateral strategy, and will contribute to the general understanding of covert articulatory variants.

**2pSCb8. Tongue root position in Hijazi arabic voiceless emphatic and non-emphatic coronal consonants.** Abdullah H. Alfaifi (Dept. of Linguist, Indiana Univ., Bloomington, IN 47405, abdalfai@indiana.edu), Malgorzata E. Cavar (Linguist, Indiana Univ., Bloomington, IN), and Steven M. Lulich (Speech & Hearing Sci., Indiana Univ., Bloomington, IN)

The longstanding interest in Arabic emphatic consonants stems in part from their articulatory variability across dialects. In different dialects, emphatics are variously described as pharyngealized, uvularized, velarized, or even glottalized consonants. This pilot study analyzes ultrasound images of the voiceless coronal emphatic consonants (traditionally transcribed as pharyngealized /tˤ/ and /sˤ/), and compares them with the corresponding non-emphatics (/t/ and /s/), and with the uvular /q/ in identical vocalic environments. We also examine coarticulatory interactions between emphatic consonants and adjacent vowels. Data from two native speakers of Hijazi Arabic showed more retracted tongue root in emphatics compared to non-emphatics. The emphatics are different from uvulars, in that the tongue dorsum is not raised toward the uvular place of articulation. Non-emphatic coronals are potentially velarized, or even palatalized, especially in the context of a front vowel. Short vowels (/a/, /i/, /u/) were more affected by emphasis in the adjacent consonant than the long vowels (/a:/, /i:/, /u:/), with more retracted tongue root position in the context of emphatic consonants. The high back vowels (/u/, /u:/) were less affected by emphasis than the other vowels.

**2pSCb9. Alveolar stops exhibit greater coarticulatory resistance than retroflexes and dentals in Malayalam.** Meghavarshini Krishnaswamy (Dept. of Linguist., Univ. of Arizona, Communications Bldg. Rm. 109, Tucson, AZ 85721, mkrishnaswamy@email.arizona.edu), Indranil Dutta (School of Lang. and Linguist, Jadavpur Univ., Kolkata, West Bengal, India), and Maumita Bhaumik (Dept. of Linguist, English and Foreign Lang. Univ., Hyderabad, Telangana, India)

Tight coronal contrasts in Malayalam geminates ($V_1$t̪:$V_2$ vs. $V_1$t̠:$V_2$ vs. $V_1$t:$V_2$) are examined in an Ultrasound study of tongue contours to understand the nature of coarticulatory resistance and aggressiveness. Degree of Articulatory Constraint (DAC) predicts that articulatory complexity mitigates the nature of coarticulatory resistance (CR) cross-linguistically. Findings from our study of Ultrasound tongue contours are contrary to the predictions of the DAC, where the expectation is that the directionality of CR and aggressiveness will be Vt̪:V < Vt̠:V < Vt:V. We find that the order of CR is Vt:V < Vt̪:V < Vt̠:V. Greater variability in tongue constriction location in retroflex and dentals is found compared to alveolars. Low neighborhood density alveolars exhibit low contextual adjustments even at the constriction. This is a consequence of greater articulatory complexity. Malayalam alveolars are distributionally restrictive compared to dental and retroflex places of articulation. Alveolars in Malayalam have low neighborhood densities which may indeed govern coarticulatory resistance and aggressiveness of this place of articulation. These findings have been corroborated in an acoustic study of Malayalam stops (Dutta *et al*., 2019). We discuss the implications of our findings for DAC and propose that sparse lexical representation coerces coarticulatory resistance in tight coronal place contrasts.

TUESDAY AFTERNOON, 8 DECEMBER 2020 3:35 P.M. TO 4:20 P.M.

## Session 2pSCc

## Speech Communication: Clinical Topics in Speech II (Poster Session)

Authors will be at their posters from 3:35 p.m. to 4:20 p.m.

### *Contributed Papers*

**2pSCc1. A large-scale comparison of the intelligibility of unit-selection and deep-neural-network parametric synthetic voices generated from dysarthric speech.** Jason Lilley (Nemours Biomedical Res., 1701 Rockland Rd., Rm. 136B, Wilmington, DE 19803, jason.lilley@nemours.org), Jolene Hyppa-Martin (Univ. of Minnesota Duluth, Duluth, MN), and H. Timothy Bunnell (Nemours Biomedical Res., Wilmington, DE)

We report on a large-scale intelligibility study of 773 listeners who transcribed semantically unpredictable stimuli generated from 4 synthetic voices: two synthetic voices, a unit-selection voice and one based on deep neural network (DNN) parametric synthesis, from the recordings of each of two dysarthric speakers. Intelligibity was calculated as the normalized phonemic edit distance (NPED) between perceived and actual transcriptions. The DNN-based voices (NPED 0.286) were significantly more intelligible ($p < .001$) than the unit-selection voices (NPED 0.316), although there was a significant interaction ($p = .034$) with the structure of the synthesized sentence. Counterintuitively, the voices generated from the more severely dysarthric speaker (NPED 0.272) were signficantly more intelligible ($p < 0.001$) than the other speaker's voices (NPED 0.329). *Post-hoc* analysis demonstrated that while the more dysarthric speaker's speech had poorer vocal quality, was measurably slower and more variable in duration, and had a smaller vowel space, this speaker also had significantly ($p > .001$) higher formant amplitudes, narrower bandwidths, and less frequency variance; she also had more careful articulation during recording. We conclude that individual differences (some within the speaker's control) can override gross measures of dysarthria in determining synthetic voice intelligibility.

**2pSCc2. Dysarthria subgroups in talkers with Huntington's disease: Free classification versus feature-constrained classification.** Daniel Kim (Vanderbilt Univ. Medical Ctr., 8310 Medical Ctr. East, Nashville, TN 37232, daniel.kim.1@vanderbilt.edu), Sarah Diehl, Michael de Riesthal, Daniel Claassen, and Antje Mefferd (Vanderbilt Univ. Medical Ctr., Nashville, TN)

In our recent study (Diehl *et al*., 2019), we examined the perceptual speech features of 48 talkers with Huntington's disease (HD) using the classic feature-based dysarthria rating scale (Darley *et al*., 1969). A cluster

analysis based on speech feature ratings revealed four dysarthria subgroups within our cohort of talkers with HD. Talkers within each subgroup shared deviant speech features that set them apart from talkers with HD in other subgroups. Presumably, talkers with similar patterns of deviant speech features sound alike. In the current study, we will test this notion using a free classification approach. Specifically, 20 naïve listeners will be asked to sort the speech samples of 48 talkers with HD that were used in the previous study into similar-sounding groups. Each listener's grouping decision will be submitted to an additive similarity tree cluster analysis to determine dysarthria subgroups based on the free classification task. Moreover, subgroup-specific speech features will be determined based on the previously established perceptual ratings of the classic dysarthria rating scale. Study outcomes will provide insights into the saliency of specific perceptual speech features in talkers with HD.

**2pSCc3. A pilot study on the acoustic characteristics of ambiguous sentences in highly intelligible individuals with Parkinson's disease.** Christina Kuo (Commun. Sci. and Disord., James Madison Univ., 235 Martin Luther King Jr. Way, MSC 4304 James Madison University, Harrisonburg, VA 22807, kuocx@jmu.edu) and Matthew Barrett (Neurology, Virginia Commonwealth Univ. School of Medicine, Richmond, VA)

Phonetic and semantic ambiguities have been documented with internal open junctures, where phonemes in the same order occur at varied linguistic boundaries and often with different prosodic patterns [Lehiste, *Phonetica* **5** (Suppl. 1), 5–54 (1960); Fisher and Logemann, *Q. J. Speech* **53**(4), 365–373 (1967)]. As such, internal open junctures offer unique opportunities for evaluating the intricate speech-language interplay [Spencer and Wollman, *Lang. Speech* **23**(2), 171i–198 (1980)]. In this study, sentences containing internal open structures were examined for individuals with Parkinson's disease (PD) of high sentence intelligibility. Given prosody and speech timing are commonly impacted in PD, it is hypothesized that internal open junctures may provide a context for detecting subtle speech changes before intelligibility is reduced. Ambiguous sentence pairs were produced by four males with the diagnosis of PD and two neurologically healthy males as a part of a larger study. Data from male speakers were studied to control for potential effects of sex in this pilot. Segment (sentence, juncture, and pause) duration, articulation rate, second formant (F2) trajectory, and fundamental frequency (F0) contour were analyzed. Specifically, the research question of interest is whether the acoustic characteristics associated with internal open junctures in PD differ from those of healthy individuals. This research is supported by a 4-VA Collaborative Research Grant (https://4-va.org/) awarded to the authors.

**2pSCc4. Are there any differences between the tongue and posterior pharyngeal wall movement patterns during normal and Masako swallow?—An ultrasound analysis.** Emily Q. Wang (Commun. Disord. & Sci., Rush Univ. Medical Ctr., 600 S. Paulina St., Ste. 1017, Chicago, IL 60612, emily_wang@rush.edu), Alison Perlman (Commun. Disord. & Sci., Rush Univ. Medical Ctr., Chicago, IL), Adam Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA), and Leonard A. Verhagen Metman (Dept. of Neurological Sci., Rush Univ. Medical Ctr., Chicago, IL)

Parkinson's disease (PD) is a neurodegenerative movement disorder affecting volitional movement due to Dopamine-producing neuron death in the substantia nigra. Impaired swallow function is highly prevalent with nearly 100% involvement in PD during the disease course and involves all three phases of swallowing. The oral stage of swallowing is impaired first. Reduced bolus control and reduced interior-to-posterior movement of the bolus transfer lead to delayed swallowing response, premature spillage, decreased base of tongue (BOT) retraction and strength, decreased hyolaryngeal elevation and excursion, and reduced action of pharyngeal constrictors. Together, these result in bolus retention, residue throughout the pharynx, and reduced airway protection leading to significant aspiration. To target these deficits we have designed an exercise regime to teach early to mid-stage PD patients to do the Masako swallow maneuver of saliva (swallow with tongue-tip held between teeth), with high intensity and frequency (i.e., 120 repetitions per day) in one month. The preliminary results are very promising. All participants demonstrated positive changes. To understand the underlying mechanism, we used Ultrasound and recorded 20 normal subjects while they did three normal swallows and 3 Masako swallow maneuvers. The initial observations indicated the tongue and pharyngeal movement patterns differ between the two maneuvers.

**2pSCc5. Articulatory strategies and their acoustic consequences: Investigating tongue retraction and lip protrusion tradeoffs in talkers with amyotrophic lateral sclerosis.** Reilly Johnson (Dept. of Hearing and Speech Sci., Vanderbilt Univ. Medical Ctr., 8310 Medical Ctr. East, Nashville, TN 37232, reilly.m.johnson@vanderbilt.edu), Nadia Kim (Vanderbilt Univ. Medical Ctr., Nashville, TN), Kris Tjaden (Univ. at Buffalo, Buffalo, NY), and Antje Mefferd (Vanderbilt Univ. Medical Ctr., Nashville, TN)

Observations of tradeoffs between lip protrusion and tongue retraction during productions of /u/ suggest that typical talkers can vary their articulatory strategies without affecting their acoustic output. In this study we tested the hypothesis that talkers with dysarthria due to amyotrophic lateral sclerosis (ALS) may take advantage of such trading relations and exaggerate lip protrusion to preserve speech acoustics in the presence of impaired tongue retraction. 14 talkers with mild to moderate dysarthria due to ALS (8 females, 6 males) and 14 age- and sex-matched controls produced "Tomorrow Mia may buy you toys again" five times at their habitual rate. Speech kinematics were recorded using electromagnetic articulography. Tongue retraction and lip protrusion were measured by calculating the displacement of the posterior tongue and upper lip sensors in the anterior-posterior dimension of the midsagittal plane during the production of "buy you." F2 transition extent was measured by calculating the change in F2 (F2 maximum-minimum) during "you." Group means of kinematic and acoustic variables were compared to determine between-group differences in articulatory and acoustic performance. Linear regressions were used to determine the contribution of each articulator as well as the combined effect of both articulators to F2 transition extent within each group.



**2pSCc6. Does asymmetry in tongue anatomy affect asymmetry in tongue position? Glossectomy and control subjects.** Maureen Stone (Univ. of Maryland Dental School, 650 W. Baltimore St. Rm. 8207, Baltimore, MD 21201, mstone@umaryland.edu), Ghaddy AlSaty, Lisa C. Honig, Joshua Lubek, Jiachen Zhuo (Univ. of Maryland Dental School, Baltimore, MD), and Jerry L. Prince (Johns Hopkins Univ., Baltimore, MD)

Glossectomy surgery removes part of the tongue due to cancer. The resulting anatomical asymmetries may affect motor symmetry. This study examines anatomical and positional tongue asymmetry in glossectomies and controls. The goals are to determine the extent of the anatomical asymmetries, and their effect on resting asymmetry. We expect that patients with unilateral resections will be more asymmetrical than controls, but that their midline tongue will be centered in the oral cavity (OC), because their dentition is unchanged and the tongue will rest in its familiar position. 3-D tongue volumes were extracted from high-resolution MRI data using Matlab. Tongue volumes were calculated for the whole tongue, both halves and the septum using ITK-SNAP. We bisected the OC in the sagittal plane from the mandibular symphysis to the center of the spinal cord. The tongue volume was calculated in each half of the OC. Results showed more anatomical asymmetry in the patients; seven patients and three controls had a volume difference of 3% or more between the left and right tongue. Positional measures showed that the septum was mostly in one half of the OC. When the septum volume was removed, the tongue volume distributed fairly equally in the OC.

**2pSCc7. Acoustic correlates of laryngeal control: Parkinson's and healthy older adults.** Marcelo S. Vieira (School of Commun. Sci. and Disord., McGill Univ., 2320 Jean Talon Est, Apt. 4, Montréal, QC H2E 1V7, Canada, marcelo.vieira@mail.mcgill.ca), Noémie Auclair-Ouellet, and Meghan Clayards (School of Commun. Sci. and Disord., McGill Univ., Montreal, QC, Canada)

Vocal fold tension is commonly used to control pulmonary airflow and subglottal pressure, resulting in a positive f0-intensity correlation. Since Parkinson's disease (PD) constrains fine movements, this mechanism may be impaired in PD. We analyzed the f0-intensity correlation in PD and a

healthy age- and gender-matched control group. For that, we extracted gender-normalized f0, intensity, and spectral emphasis (SE) from each syllable in three sentences of a read text. Additionally, from a sustained [a] task, we measured maximum phonation time ([a] duration; MPT) as well as jitter and shimmer (combined using PCA; JS). Using Linear Mixed Models, we confirmed the f0-intensity correlation in each group. Furthermore, JS interacted with intensity, indicating that voice instability weakens the correlation. No MPT effect was found. Importantly, even controlling for JS and MPT, the f0-intensity correlation was significantly weaker in PD. Lastly, we build a model using SE instead of intensity and only a negative correlation was found. Overall, this study suggests that voice instability negatively affects airflow control, but is not sufficient to explain its reduction in PD. Moreover, it indicates that the SE-f0 relationship is preserved in PD and it is not affected by the voice parameters analyzed.

**2pSCc8. Acoustic characteristics of conversational and clear speech produced in quiet and in noise by older adults with hearing loss.** Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, 390 South, 1530 East, Rm. 1201, Salt Lake City, UT 84112, sarah.ferguson@hsc.utah.edu), Paige D. Morisak, Sara Thurston, Kylie N. Willson (Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT), and Eric Hunter (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI)

When communication occurs in noise or with hearing-impaired listeners, talkers often modify their speech. Speech produced in noise, or "Lombard speech," and speech produced by talkers instructed to speak as though their communication partner has a hearing loss, or "clear speech," usually feature a reduced speaking rate and a higher and more variable voice pitch than quiet and/or conversational-style speech. Both styles are also generally more intelligible than quiet/conversational speech. Although clear speech in particular is meant to accommodate the needs of listeners with hearing loss, most of the talkers in previous studies have been young adults with normal hearing. We hypothesize that talkers with hearing loss, having experience as listeners with hearing loss, would better know what listeners with hearing loss need than talkers with normal hearing, and that the magnitude of the acoustic differences would change accordingly. In the present study, talkers aged 65–85 years with bilateral mild-to-moderately severe sloping sensorineural hearing loss were recorded reading sentences in quiet and in the presence of white noise under instructions to speak conversationally and then clearly. Acoustic differences between the styles for these talkers will be compared to those observed previously in young normal-hearing adults using the same recording protocol.

**2pSCc9. Diadochokinetic speech in individuals at clinical high risk for schizophrenia.** Kasia Hitczenko (Northwestern Univ., 2000 Sheridan Rd., Evanston, IL 60208, kasia.hitczenko@northwestern.edu), Yael Segal, Tzeviya Sylvia Fuchs (Bar Ilan Univ., Ramat Gan, Israel), Matthew Goldrick (Northwestern Univ., Evanston, IL), Joseph Keshet (Bar Ilan Univ., Ramat Gan, Israel), and Vijay Mittal (Northwestern Univ., Evanston, IL)

Past work has shown that individuals at clinical high risk (CHR) for developing psychosis have difficulty with motor coordination (e.g., Mittal *et al.*, 2013). Here we ask whether these motor disruptions also affect their ability to produce speech, resulting in more variable speech productions. CHR participants and matched controls completed a diadochokinetic speech task, in which they first repeated syllables (i.e., papap…, tatat…, kakak…) and then alternated between different syllables (i.e., pataka....) as quickly and accurately as possible. We detect and localize the acoustic syllables /pa/ , /ta/ and /ka/ automatically using a deep learning algorithm that was specially designed for the detection of 'acoustic objects', discrete events in the acoustic signal (Segal *et al.*, 2019). We predict that, due to less stable control of their articulators, CHR individuals will show increased acoustic variability and more overlap between different sound categories relative to healthy controls.

**2pSCc10. An acoustic study of plosives production in Cantonese speakers with Parkinson's disease.** Yiting Chen (Dept. of Chinese and Bilingual Studies, The Hong Kong Polytechnic Univ., 11 Yuk Choi Rd., Hung Hom, Kowloon, Hong Kong, Hong Kong, yiting.chen@polyu.edu.hk), Min Ney Wong, Crystal Tze Ying CHOW, Xuan Wang, Yuhan Lin (Dept. of Chinese and Bilingual Studies, The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong), Manwa L. Ng (Speech and Hearing Sci., Univ. of Hong Kong, Pokfulam, Hong Kong), and Shirley Yin Yu Pang (Dept. of Medicine, Queen Mary Hospital, Hong Kong, Hong Kong)

Articulatory impairment, including consonant imprecision, has been widely studied in patients with Parkinson's disease (PD). However, acoustic investigation of articulatory impairment in Cantonese speakers with PD has been limited. The purpose of this study was to investigate the acoustic characteristics of plosives production in Cantonese speakers with PD. The participants comprised 17 Cantonese speakers with PD and 17 age- and gender-matched healthy controls (HC). Six Cantonese plosives /p, $p^h$, t, $t^h$, k, $k^h$/ followed by the vowel /a/ were produced at high-level tone (T1) in the context of word and sentence. All speech samples produced by PD patients were further divided into two subgroups: normal plosive production (PD-NP) and spirantized plosive production (PD-SP). Higher intensity ratio, shorter VOT and shorter closure duration during plosive production were found in both PD-NP and PD-SP subgroups when compared to the HC group. In addition, aspiration and context did affect the intensity ratio, VOT and closure duration while place of articulation only affected the VOT. Furthermore, it was found that the most commonly misarticulated plosives in Cantonese speakers with PD were bilabial stops, followed by alveolar and velar stops. This finding of spirantization of plosives is in agreement with previous studies.

**2pSCc11. Acoustic-phonetic analysis of speech following a lobectomy or esophagectomy.** Babar Khan (Indiana Univ. School of Medicine, Regenstrief Inst., Indianapolis, IN), Michael Stokes (Waveform Commun., LLC, 3929 Graceland Ave., Indianapolis, IN 46208, mstokes@waveformcommunication.com), Sikandar Khan, and Sarah Seyffert (Indiana Univ. School of Medicine, Regenstrief Inst., Indianapolis, IN)

An observational study was conducted to monitor and detect the onset of altered level of consciousness and delirium in patients after a major non-cardiac thoracic surgery. Speech recordings from patients were recorded pre-operatively and once daily in the post-operative setting until hospital discharge. A vocabulary consisting of vowels in a coarticulatory neutral environment is used to identify differences in timing or typical articulatory movements that would be indicative of impairment due to delirium. More than one recording session across days is needed in order to compare the patients' speech to their own speech to identify patterns and changes for that talker. Programming has been developed that utilizes the Praat software package to take acoustic-phonetic measurements of pitch, formant frequencies, and amplitude every 6 ms. This processing has generated tens of thousands of rows of data across 1,166 h-vowel-d words reflecting typical patterns for each of the 16 subjects with more than one recording session. This unique dataset has led to insights that go beyond the original intent of identifying the potential onset of delirium including patterns of dyspnea resulting from the surgery. The initial findings from mining the extensive dataset will be presented.

## Session 2pSPa

# Signal Processing in Acoustics, Animal Bioacoustics, Engineering Acoustics, Underwater Acoustics, and Acoustical Oceanography: Acoustic Localization I

Zoi-Heleni Michalopoulou, Cochair
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Paul J. Gendron, Cochair
*ECE, Univ. of Massachusetts Dartmouth, 285 Old Westport Rd, Dartmouth, MA 02747*

**Chair's Introduction—1:05**



### *Invited Papers*

**1:10**

**2pSPa1. Battlefield acoustics: Applications and challenges.** W. C. K. Alberts (CCDC-Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783, william.c.alberts4.civ@mail.mil)

Some primary objectives of research and development for battlefield acoustic sensing include the optimization of sensors, arrays, and signal processing algorithms for the detection, localization, and classification of diverse broadband and transient threats in complex environments. Time-difference-of-arrival and frequency domain signal processing techniques using distributed microphone arrays allow sound source localization and tracking through triangulation in the 3-D space. This presentation will introduce general applications of broadband acoustic and infrasonic microphone arrays, acoustic particle velocity sensors, geophones, propagation environment effects and wind mitigation methods.

**1:30**

**2pSPa2. General formulation of source localization for multiple receivers with fully saturated scattered signals.** D. Keith Wilson (U.S. Army Engineer Res. and Development Ctr., U.S Army ERDC-CRREL, 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil), Chris L. Pettit (Aerosp. Eng., U.S. Naval Acad., Annapolis, MD), Vladimir Ostashev, and Matthew J. Kamrath (U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

A general formulation is provided for the problem of localizing a source from signals at multiple, arbitrarily separated receivers. The signals are assumed to be narrowband and fully saturated by scattering from homogeneous turbulence as they propagate along line-of-sight paths to the receivers. In this case, the complex Wishart distribution can be used to describe the joint distribution of the received signals. Previously derived results from the theory of wave propagation in random media can furthermore be used to model the covariance matrix between the received signals. Then, given a prior distribution on the source location, Bayes's theorem enables the maximum a posteriori (MAP) estimate for the source location to be formulated. This general approach incorporates source localization by beamforming and by signal amplitude as special cases. The behavior of the solution is illustrated with numerical examples.

**1:50**

**2pSPa3. Propagation of audio signals from elevated harmonic sources.** Geoffrey H. Goldman (U.S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783-1197, geoffrey.h.goldman.civ@mail.mil)

The propagation of signals from elevated sources in the lower audio band (30–500 Hz) are simulated and processed to obtain angle of arrival (AOA) estimates using a small acoustic array. The propagation of the signals are simulated using ray tracing through a stratified atmosphere that is implemented with a finite difference time domain method described by Pierce. The wind and temperature profiles are modeled using Monin-Obukhov similarity theory. The ground reflection is modeled using a two-parameter ground impedance model developed by Attenborough. The spatial coherence of the signals at the array are handled using a statistical model developed by Kozick *et al*. Basically, the coherence at difference microphones is reduced based upon the range of the source and distance between microphones. The source is modeled using a harmonic structure with no atmospheric or spherical attenuation. This is reasonable since the amplitude of sources is often not known. The AOA of the simulated signals are estimated using a beamforming algorithm and analyzed.

2:10

**2pSPa4. Sound source localizations using various microphone arrays.** Ning Xiang (School of Architecture, Rensselaer Polytechnic Inst., 110 Eighth St., Troy, NY 12180, xiangn@rpi.edu)

Estimating directions of arrival (DoAs) of multiple sound events still posts challenges in acoustic source localization. This paper reviews recent development on model-based direction-of-arrival analysis using three different microphone array configurations, in situations involving potentially multiple concurrent sound sources. The array configurations include a sparse arrangement consisting of two microphones, coprime linear microphone arrays, and spherical microphone arrays. Formulation and evaluation of prediction models are of central importance in coping with these challenges. These three array configurations are all used to solve a two-level inferential problem of sound source enumeration and direction of arrival estimation. The prediction models based on specific array configurations are evaluated upon experimental data derived from pertinent array processing in order to select the simplest such model that can adequately describe the experimental data, thereby enumerating first the concise number of sound sources, then their DoA information. This paper reviews the prediction models recently established for the three microphone arrays, the two levels of inference, and analysis results. This paper will highlight the importance of the prediction models for the model-based analysis of complex sound environments with potentially multiple concurrent sources.

TUESDAY AFTERNOON, 8 DECEMBER 2020 1:05 P.M. TO 2:35 P.M.

## Session 2pSPb

## Signal Processing in Acoustics: Knowledge Discovery and Information Representation for Signal Processing in Acoustics III

Ananya Sen Gupta, Cochair
*Department of Electrical and Computer Engineering, University of Iowa, 103 S Capitol Street, Iowa City, IA 52242*

Benjamin N. Taft, Cochair
*Landmark Acoustics LLC, 1301 Cleveland Ave., Racine, WI 53405*

**Chair's Introduction—1:05**

### *Contributed Papers*

1:10

**2pSPb1. Representing sonar target spectral features using a two-dimensional Gabor wavelet.** Bernice Kubicek (Elec. and Comput. Eng., Univ. of Iowa, 103 South Capitol St., Iowa City, IA 52242, bernice-kubicek@uiowa.edu), Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), and Ivars Kirsteins (Naval Undersea Warfare Ctr., Newport, NJ)

We present an alternate representation of sonar spectral features with popular machine learning techniques for automated target classification. Sonar target recognition suffers from feature uncertainties due to unpredictable environmental parameters: changing sound speed profiles, inhomogeneous propagation of acoustic waves, interference and delayed echoes due to multipath effects, ambient ocean noise and reverberation, and signal deformation due to reflections from the seafloor, sea surface, and the Doppler effect. These parameters are further dependent on target orientation and may combine in a nonlinear fashion, making target classification near impossible. We propose a representation of sonar spectral features using the two-dimensional Gabor wavelet as a kernel filter. This extracts highly informative features specific to a target, regardless of orientation. We validate the robustness of our representation against feature uncertainties by comparing the classification performance of three machine learning techniques—a support vector machine, random forest tree, and a neural network—trained on the raw spectrogram (acoustic color) versus the Gabor-filtered feature space through resulting confusion matrices. We will provide classification results of public domain experimental field data. [This research is funded by the Office of Naval Research under Grant No. N00014-19-1-2436.]

1:30

**2pSPb2. Braid representation of the shallow water acoustic channel to interpret temporal evolution of multipath arrivals.** Ryan A. McCarthy (Dept. of Elec. and Comput. Eng., Univ. of Iowa, 103 S. Capitol St., Iowa City, IA 52242, ryan-mccarthy-1@uiowa.edu) and Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA)

We present a morphological representation of the underwater acoustic channel that employs geometric braids to gain real-time knowledge of the dominant multipath activity. The key idea is to incorporate braid manifolds in tandem with non-convex mixed norm optimization techniques to track the temporal evolution of multipath arrivals. Specifically, we focus on channel braids that manifest as rapidly fluctuating high-energy taps in the delay spread as well as the delay-Doppler scattering function. Multiple braids can also be topologically combined using braid operations to interpret oceanic phenomena such as caustics and surface wave focusing, among others. We present techniques for adaptively updating the channel braids and their overlap patterns to reflect the temporal evolution of the shallow water acoustic channel. We evaluate the performance of proposed morphological channel estimation technique in terms of the normalized prediction error and computational time. Results based on numerical channel simulations based on diverse oceanic and experimental conditions as well as experimental field data from the SPACE08 experiment will be presented.

**1:50**

**2pSPb3. Dimensionality reduction of cross-spectral density matrices using diffusion map projections.** Steven I. Finette (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375-5320, steven.finette@nrl.navy.mil)

Cross-spectral density matrices (CSDMs) constructed for matched-field source localization in an ocean waveguide are both Hermitian and positive semi-definite, allowing them to be interpreted in a non-Euclidean geometric framework as "points" in a Riemannian manifold. This abstract matrix representation can be used to construct distance metrics on the manifold, and distances between pairs of such matrices provide a measure of similarity between the true source CSDM and various replica CSDMs. With this geometric interpretation, the shortest "distance" between pairs of source/replica CSDMs represents an estimate of the source location obtained from acoustic field amplitude and phase acquired on a vertical array [Finette and Mignerey, *JASA* **143** (2018)]. In this presentation, visualizations of CSDM manifolds obtained from simulated acoustic fields propagating in an ocean waveguide with internal wave-induced variability are illustrated to gain insight into this approach to passive source localization. Since the original manifold resides in a high-dimensional space determined by the number of sensors, manifold learning using diffusion maps is employed to reduce the dimensionality but constrained to retain the relative distance relationships among the CSDMs. The original manifold is projected down to three dimensions for visualization. [Work supported by the Office of Naval Research.]

## *Invited Paper*



**2:10**

**2pSPb4. A tiered, bio-inspired, modular framework for robust acoustic feature extraction.** Benjamin N. Taft (Landmark Acoust. LLC, 1301 Cleveland Ave., Racine, WI 53405, ben.taft@landmarkacoustics.com)

Vertebrates separate audition into peripheral, brain-stem, and fore-brain stages. These stages are anatomically and theoretically separable, and our understanding of hearing has improved by analyzing them as such. Here, I outline a strategy for structuring acoustic feature extraction that is inspired by the tiered structure of the brain. The system aims to work online in close to real time, using a modest amount of memory and processing power. Another goal is to maintain a domain-appropriate conceptual connection between the features that we are measuring and the physical and/or biological processes that we are monitoring. We do this by breaking feature extraction into stages based on context. Some features will be useful across many contexts, or useful for distinguishing among contexts. We should extract those features at each time step of our analysis. Other features will only be relevant within specific contexts, so we should only spend computational resources on extracting them when they are relevant. This tiered computational strategy yields multiple points of introspection about the quantitative relationships between continuous features and discrete contexts. We can use quantitative introspection to analyze how well our modules generalize to other contexts, and to validate empirical models of how those contexts work.

# Session 2pSPc

## Signal Processing in Acoustics, Animal Bioacoustics, Engineering Acoustics, Underwater Acoustics, and Acoustical Oceanography: Acoustic Localization IV

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**Chair's Introduction—2:50**

### *Contributed Papers*

**2:55**

**2pSPc1. Implementation of a microphone array for improved noise path estimation in a feed-forward active noise control system.** Heui Young Park (Mech. Eng., The Cooper Union for the Advancement of Sci. and Art, 41 Cooper Square, New York, NY 10003, park3@cooper.edu), Daniel Abes, Martin S. Lawless, and Dirk M. Luchtenburg (Mech. Eng., The Cooper Union for the Advancement of Sci. and Art, New York, NY)

Active noise control (ANC) systems detect incoming sound and generate an anti-noise signal to attenuate it through destructive interference. ANC systems are generally limited to low-frequency sounds because the sensors and actuators are close together, leaving insufficient time to react to high-frequency or impulsive sounds. To compensate, common applications, such as noise-cancelling headphones, rely on complete ear coverings to block out higher-frequency sound indiscriminately, potentially hindering communication and situational awareness. The present work proposes to use an external array of microphones surrounding the user to increase the distance between the sensors and the user to afford more time to generate anti-noise signals and provide better path estimation. Using time delay of arrival (TDOA), the array tracks the locations of the user and the noise sources in real-time. The ANC system uses a feed-forward Filtered-x Least Mean Squared (FxLMS) algorithm that adjusts the weights of the controller based on the TDOA path estimation instead of adapting the filters with an error microphone and feedback loop. Simulations of the proposed system and feedback FxLMS path estimation were conducted in MATLAB. Compared to the feedback FxLMS algorithm, the TDOA system yielded $11 \pm 1$ dB less root-mean-square error in the generated anti-noise signals.

**3:15**

**2pSPc2. Low-latency localization in the spherical harmonics domain using an iterative search method.** Jonathan Mathews (Architectural Sci. - Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Greene Bldg., Troy, NY 12180, mathej4@rpi.edu) and Jonas Braasch (Architectural Sci. - Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Higher-order ambisonic microphones have attained considerable interest for their ability to allow spherical harmonic decomposition of soundfields. In taking advantage of this property, myriad methods have been produced for directional filtering and subsequent analysis, as well as localization and tracking of soundfields. This study introduces a technique to estimate direction of arrival of an acoustic signal by generating a set of spatial filters corresponding to randomly chosen orientations over the sphere, then iteratively reducing the region for orientation selection based on the maximum power obtained from the filter response. This method is distinct from existing mapping- and vector-based methods in that it produces accurate, low-latency results in the presence of moderate reverberation. Results obtained using both simulated and experimental data demonstrate viability of this technique for localization in large, reverberant spaces as well as accurate source identification under real-time conditions. [Work supported by NSF grants #1631674 and #1909229, and the RPI Cognitive and Immersive Systems Laboratory.]

**3:35**

**2pSPc3. Improving the performance of the Bartlett method for single-snapshot direction-of-arrival (DOA) estimation using signal processing on graphs (SPG).** Eldridge Alcantara (Elec. & Comput. Eng., Univ. of Washington, Seattle, Campus Box 352500, Seattle, WA 98195, eealcant@uw.edu), Shima Abadi, and Les Atlas (Elec. & Comput. Eng., Univ. of Washington, Seattle, Seattle, WA)

Single-snapshot direction-of-arrival (DOA) estimation has long been studied under the framework of conventional signal processing. The most well-studied method estimates DOA from the power spectrum of data that is assumed to lie on a fixed grid in which samples are uniformly spaced, an algorithm known as the conventional Bartlett method. By using a fixed grid, the linear weights used to calculate power are fixed and the method's estimation performance, which can be measured by RMSE versus SNR, is fixed as well. Is it possible improve the Bartlett method's performance by appropriately adjusting these linear weights? In this presentation we show that we can if we apply principles and tools from Signal Processing on Graphs (SPG) so that data now lies on a variable grid known as a graph. We first replicate the system used for the conventional Bartlett method using SPG and then extend the system so it can work on other grids (or graphs). From this SPG-based system, we then show there exist graph structures that will produce linear weights in which RMSE, for certain SNRs, are lower than the conventional Bartlett method. Finally, we validate these results further through simulations.

**3:55**

**2pSPc4. Determining the range to marine mammals in the Northern gulf of mexico via bayesian acoustic signal processing.** Kendal Leftwich (Phys., Univ. of New Orleans, 1021 Sci. Bldg., New Orleans, LA 70148, kmleftwi@uno.edu), George Drouant (Oregon Inst. of Technol., Klamath Falls, OR), and Juliette W. Ioup (Phys., Univ. of New ORleans, New Orleans, LA)

Being able to determine the range or distance to an object from its acoustic signal is a key component in tracking the object. We will apply concepts from the Bayesian processor (Kalman filter) as well as known environmental aspects to LADC-GEMM underwater passive acoustic data: sperm whales, beaked whales and dolphins in the northern Gulf of Mexico. The Bayesian processor has had applications in chemical processing, navigation, ocean acoustics, seismology, and tracking, among others. We will use a modified version of the Kalman filter with data from a single hydrophone to determine the most-probable range of the marine mammal from the hydrophone. We will use such aspects as the geometry of the region, the depth of the hydrophone, the range of possible detection, the structure of the click and other factors to restrict the possible locations of the marine mammal source. From this, we can determine a three-dimensional shell of most probable locations for the marine mammal.

TUESDAY AFTERNOON, 8 DECEMBER 2020 1:05 P.M. TO 2:35 P.M.

## Session 2pUWa

## Underwater Acoustics: Detection and localization in the Ocean I

Martin Siderius, Chair
*Portland State Univ., 1600 SW 4th Avenue, Suite 260, Portland, OR 97201*



**Chair's Introduction—1:05**

***Contributed Papers***

**1:10**

**2pUWa1. Data-driven source localization using shipping sources of opportunity.** Nicholas C. Durofchalk (Mech. Eng., Georgia Inst. of Technol., 2788 Defoors Ferry Rd., Apt. 325, Atlanta 30318, Georgia, ndurofchalk3@gatech.edu), Jihui Jin (Elec. and Comput. Eng., Georgia Inst. of Technol., Atlanta, GA), Priyabrata Saha (Comput. Sci., Georgia Inst. of Technol., Atlanta, GA), Justin Romberg (Elec. and Comput. Eng., Georgia Inst. of Technol., Atlanta, GA), Saibal Mukhopadhyay (Comput. Sci., Georgia Inst. of Technol., Atlanta, GA), and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Underwater source localization is often achieved with a purely model-based approach such as matched-field processing, which relies on simulated replica-field. However, such approaches only yield reasonable predictions if the complex and dynamic ocean environment is sufficiently known – often a daunting task. Alternatively, it has been suggested that channel impulse responses (CIRs) estimated from measurements of sources of opportunity (such as commercial shipping vessels) can feed a data-driven approach to source localization that forgoes the need for precise model-parameters [Durofchalk *et al.*, *JASA* **146**(4), 2691–2691 (2019)]. In this presentation, vertical line array (VLA) data from the SBCEx16 experiment conducted in the vicinity of shipping lanes in the Santa Barbara channel (580 m depth, downward refracting profile) are first used to a construct a library of estimate CIRs between selected locations along shipping tracks and VLA receivers using ray-based blind deconvolution (RBD) [Byun *et al.*, *JASA* **141**(2), 797–807 (2017)]. Subsequently, this library of data derived CIRs is used to train a machine learning algorithm to localize other surface sources. The average localization error and computational efficiency obtained with this machine learning approach is compared to that of traditional matched field processing techniques.

**1:30**

**2pUWa2. Quantifying the performance of ray-based Blind Deconvolution (RBD) algorithm for long-range shipping sources.** Richard X. Touret (Ocean Sci. and Eng., Georgia Inst. of Technol., 127 Sisson Ave. NE, Atlanta, GA 30317, rtouret@gatech.edu), Nicholas C. Durofchalk, and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

The ray-based Blind Deconvolution (RBD) can provide an estimate of the relative arrival-times of the channel impulse responses (CIRs) between a shipping source (acting as a shallow source of opportunity) and a remote receiver array (e.g., a short bottom-mounted vertical line array (VLA)) [Durofchalk and Sabra, *JASA* **147**(3), 1927–1938 (2020)]. Previous RBD studies primarily rely on beamforming on the (most energetic) direct path to estimate the phase of the unknown source; but in typical downward refracting environments the direct path is only observable at short ranges for a (very shallow) shipping source. Alternatively, the overall performance of the RBD algorithm when treating long range sources of opportunity remains to be evaluated in similar settings when only bottom-interacting multipath arrivals can reach the VLA, i.e., beyond the range when the direct path is observable. In this study, a combination of numerical simulations and experimental validation (using the SBCEx16 experiment conducted in the Santa Barbara shipping channel with short, bottom-mounted VLAs) is used to investigate the maximum detection range of shipping sources of opportunity via multipath arrivals as a function of frequency, bathymetry effects and anisotropy of the shipping source and the implications for estimating the CIRs using the RBD method.

**1:50**

**2pUWa3. Acoustically measured *transmission loss* improves effectiveness of sonar search.** Charles H. Wiseman (Peninsula Publishing, 1630 Post Rd. East, Unit 312, Westport, CT 06880, chaswiseman30@gmail.com)

The *Sonar Equation* is the heart of the process for predicting the effectiveness of an Antisubmarine Warfare (ASW) ship's sonar in the underwater detection of submarines. The equation's determination of Ranges associated with probabilities of detection and false alarms is the fundamental input for the selection of the most effective sonar modes in an individual ship and, at the broader tactical scale, in the selection of ASW search plans. The Sonar Equation's predictions of sonar performance, however, are subject to considerable error when employed tactically at sea in ASW operations. A major source of the error is the input, *Transmission Loss,* usually derived from XBT "snapshots" of temperature versus water depth. The resulting predictions of sonar performance are projected *uniformly* over space and time, leading to predictions that are stale and inaccurate. Accuracy of the Sonar Equation's output, however, can be improved by measuring Transmission Loss *acoustically* and *continuously*. A stream of pulses, periodically broadcast from a specially designed expendable sound source, is measured at the front end of the ship's sonar. A large population of Transmission Loss values is built up from which strong statistical inferences regarding detection can be drawn. The power of a continuously updated population of Transmission Loss values leads to more accurate inferences of the local acoustic environment.

**2:10**

**2pUWa4. Enhanced automated recognition of underwater mine-like objects through environmentally adaptive fusion of detectors, features, and classifiers.** T. Scott Brandes (BAE Systems, FAST Labs, 4721 Emperor Blvd., Ste. 330, Durham, NC 27703, tsbrandes@gmail.com) and Brett Ballard (BAE Systems, FAST Labs, Durham, NC)

The complexity of the natural underwater environment creates a challenging arena in which to find underwater mines. In this work, we demonstrate that automated mine-like object detection tasks are greatly facilitated by a comprehensive fusion process. Our approach begins with characterization of the seafloor based on textures within synthetic aperture sonar (SAS) imagery and uses this to exploit information from the available sensors, multiple detector types, measured features, and target classifiers, to facilitate mine-like object recognition. Our approach is able to adapt as environmental characteristics change, including the ability to recognize novel seabed types. We then adaptively retrain classifiers through active learning in these novel seabed types resulting in improved mitigation of challenging environmental clutter as it is encountered, and develop a segmentation constrained network (SCN) algorithm which enables increased generalization abilities for recognizing mine-like objects in both under-represented and novel, unseen environments in available training data. Additionally, we present a fusion approach that allows us to combine multiple detectors, feature types spanning both measured expert features and deep learning, and an ensemble of classifiers, for the particular seabed mixture proportions measured around each detected target. [Work supported by the Office of Naval Research.]

TUESDAY AFTERNOON, 8 DECEMBER 2020 — 2:50 P.M. TO 4:20 P.M.

## Session 2pUWb

## Underwater Acoustics: Detection and Localization in the Ocean II

Martin Siderius, Chair
*Portland State Univ., 1600 SW 4th Avenue, Suite 260, Portland, OR 97201*

**Chair's Introduction—2:50**

### *Contributed Papers*

**2:55**

**2pUWb1. High frequency source depth localization using deep neural network.** Seung Hyun Yoon (Seoul National Univ., 1, Gwanak-ro, Gwanak-gu, 34-305, Seoul 08826, South Korea, justin1128@snu.ac.kr), Haesang Yang, and Woojae Seong (Seoul National Univ., Seoul, South Korea)

Localization of high-frequency sources in underwater is a difficult problem because the propagation model is increasingly sensitive to environmental parameters as the frequency increases. A deep learning approach is proposed to estimate the depth of a high-frequency underwater source using a single hydrophone trained on real data. A residual neural network is trained by a spectrogram of measured signal and estimates the depth of the source as a regression problem. The method is applied to data collected during the shallow water acoustic variability experiment 2015 in the northeastern East China Sea and compared with the results of the frequency difference matched field processing which uses 16 sensors. It was also validated that the trained model can display the generalized results for the signals measured at other times or at similar settings but different source-receiver location. As a result, the neural network is able to more accurately estimate the depth of high-frequency source and shows the features found by the network from real data are effective for localization, while the model often fails to generate accurate replicas.

**3:15**

**2pUWb2. Study of low frequency flight recorder detection.** I Yun Su (Eng. Sci. and Ocean Eng., National Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Taipei 10617, Taiwan, r07525010@ntu.edu.tw), Wen-Yang Liu (Eng. Sci. and Ocean Eng., National Taiwan Univ., Taipei, Taiwan), Li-Chang Chuang, Kai-Hong Fang (Taiwan Transportation Safety Board, New Taipei, Taiwan), and Chi-Fang Chen (Eng. Sci. and Ocean Eng., National Taiwan Univ., Taipei, Taiwan)

A flight recorder is installed in every aircraft to record the flight status. When the plane crashes into the ocean, the underwater locater beacon (ULB) inside the flight recorder will be triggered and make a pulse sound at least 30 days. In order to find it as soon as possible, it is important to fully understand the signal of the ULB and how it propagation underwater. Recently, low frequency flight recorder is developed to improve the detection range. The signal frequency of the ULB is not only 37.5 kHz, but also can be 8.8 kHz. For the 8.8 kHz signal has longer detection ranges due to lower transmission loss than the 37.5 kHz signal has. This study will show the simulation based on Gaussian Beam Model, the experiment results of the 8.8 kHz beacon, and the comparison of simulation and experiment results between the two different frequencies. Analysis of the detection range prediction of the 8.8 kHz beacon is also carried out in the seas surrounding Taiwan, which can be used to assist search planning of the flight recorder. [This study was sponsored by Taiwan Transportation Safety Board under Project No. NTUUAL-2020-01.]

**3:35**

**2pUWb3. Localization correction method for underwater sensors network drift with single float-anchor node.** Chen Zhao (Harbin Eng. Univ., Harbin Eng. University, Harbin, Heilongjiang 150001, China, zhaochen212930304@126.com), Feng Zhou, and Gang Qiao (Harbin Eng. Univ., Harbin, Heilongjiang, China)

Distributed localization method based on wireless sensor networks (WSNs) have been widely used in marine target locating. Main aim of underwater localization is to overcome the influence on acoustic channel and underwater environment to provide a higher accuracy for target exploration and navigation. Unfortunately, affected by the ocean current, sub-anchor nodes in WSNs will drift away from the initial position and led an obvious positioning error for the target. To overcome this difficulty, the paper proposes a novel and efficient method based on one extra float-anchor node (which the precise location is known) to optimize the positioning results. The proposed method consists of three stages: (a) to estimate the location of float-anchor node with sub-anchor nodes; (b) to compare the estimation result and real location of the float-anchor node to calculate the compensation function; (c) to locate the target with sub-anchor nodes and correct the result with compensation function. Simulation experiments clearly show that the proposed method have a significant improvement in the localization accuracy. The paper achieves correction localization method with existed sub-anchor nodes and only one extra float-anchor and obviously improve the positioning accuracy.

**3:55**

**2pUWb4. Research on detection method of single-beam echo sounder based on equivalent measurement.** Chao ChenHEN (College of Marine Sci. and Technol., Zhejiang Ocean Univ., Room 223,No. 1,Haidanan Rd., Lincheng, Zhoushan, Zhejiang 316000, China, chaochen@autosubsea.com)

In order to detect the accuracy index of the full range of the single-beam echo sounder in the laboratory sink, a detection method based on the equivalent reflection interface of the acoustic signal is proposed. A set of measurement and detection device consisting of underwater acoustic response device, laser rangefinder (standard device) and five-sided muffler pool is also designed. The acoustic signal equivalent interface adopts a DSP-based circuit design. By equivalent simulating a long-distance echo signal with the delay and attenuation of the electrical signal, different equivalent depths are constructed in the anechoic environment. Finally, a data traceability system for a single- beam echo sounder can be established through correspondence formula derivation and data analysis.

# OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday and Thursday. Most committee meetings will start at 4:30 p.m. EST, some committees choosing a later time as noted in the list below.

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:**

Acoustical Oceanography
Animal Bioacoustics
Architectural Acoustics
Engineering Acoustics
Physical Acoustics
Psychological and Physiological Acoustics
Signal Processing in Acoustics (6:30 p.m.)
Structural Acoustics and Vibration

**Committees meeting on Thursday are as follows:**

Biomedical Acoustics
Computational Acoustics
Musical Acoustics
Noise (6:00 p.m.)
Speech Communication
Underwater Acoustics

## Session 3aAAa

## Architectural Acoustics, Noise, Signal Processing in Acoustics and Engineering Acoustics: Session in Memory of Jiri Tichy I

Victor W. Sparrow, Cochair
*Penn State, 201 Applied Science Bldg., University Park, PA 16802*

Gary W. Elko, Cochair
*mh acoustics, 25A Summit Ave., Summit, NJ 07901*

**Chair's Introduction—9:30**

### *Invited Papers*

**9:35**

**3aAAa1. Jiri Tichy's contributions to acoustics and noise control engineering.** George C. Maling (INCE Foundation, 102 Acorns Way, Brunswick, ME 04011, maling@alum.mit.edu)

In the mid 1960s, my interest in reverberation rooms for the determination of sound power led me to get to know Jiri Tichy because he had similar interests in the field. At IBM, we had a good relationship with Professor Tichy and others at Penn State in making recommendations for student projects and following through with funding. He served as president of the Institute of Noise Control Engineering of the USA (INCE-USA) and was very active in presenting at the INTER-NOISE and NOISE-CON series of international and national conferences on noise control engineering. He played a major role in bringing INTER-NOISE to Prague in 2004 Four of us, William Lang, Matt Nobile, Jiri Tichy, and me also collaborated on the handbook article on determination of sound power of noise sources in the book Noise and Vibration Control Engineering edited by Leo Beranek and Istvan Ver. While his main interests were architectural acoustics and active control, he made many contributions to noise control engineering.



**9:55**

**3aAAa2. Jiri Tichy and his impact on the field of active noise control.** Scott D. Sommerfeldt (Brigham Young Univ., N249 ESC, Provo, UT 84602, scott_sommerfeldt@byu.edu)

Jiri Tichy was integrally involved in a number of cutting-edge research areas in acoustics, which gave him keen insights as to where meaningful contributions to the field of acoustics could be made. These insights and connections enabled him to successfully mentor numerous students in these areas by involving them in significant research and introducing them to key contacts. One such area was the field of active noise control, where he was both a mentor and friend to the author. Jiri became actively involved in active noise control in the 1980s, at the time when active control was reborn in the digital age and began to explode as a research area. During this time, the author had the privilege of working with Jiri in this field and being mentored by him. This paper will overview some of the research by the author that was either directly or indirectly influenced by Jiri in his role as both mentor and colleague. Two such research areas include the technique of passive real-time system identification, and the method of actively minimizing acoustic energy density as a means of achieving more global attenuation of the sound field.

**10:15**

**3aAAa3. Learning architectural acoustics at Penn State with Jiri Tichy 1971 to 1977.** Kenneth P. Roy (LeShanShui Consulting LLC, 136 Magnolia Dr., Holtwood, PA 17532, kenneth.p.roy@gmail.com)

While studying electrical engineering at the University of Maine, I became aware of the specialize field of "acoustics." After graduation from UMaine with no specific plan in sight, it occurred to me that graduate studies in architectural acoustic would be of interest. So, I wrote to Leo Beranek to inquire about which schools might be considered for such graduate work. From Leo's list of five schools I applied to two, one of which offered an assistantship, and Penn State which did not. I chose Penn State because of a professor in Architectural Engineering—Jiri Tichy. After the first semester, I was offered a graduate assistantship doing research on small room acoustics directly with Jiri Tichy, and I was his first student to receive an M.S. degree in acoustics through the Dept. of Architectural Engineering in 1973. I changed course along with Jiri as he became the director of the Interdisciplinary program in Acoustics, and I continued with his research assistantship graduating with a Ph.D. in Acoustics in 1977. I retired from a career in research in 2017 having spent my entire life doing what I learned because of Jiri, both in technology and in life.

**10:35**

**3aAAa4. Research on active noise control and intensity with Professor Jiri Tichy 1984–2000.** David C. Swanson (Penn State ARL, 222E ARL Bldg., PO Box 30, State College, PA 16804, dcs5@psu.edu)

Before Widrow [1] published the filtered-x algorithm in 1985, Warnaka *et al*. had a working prototype of a feedforward ANC which also pre-filtered the error path [2]. Tichy's role in this early patent was to break down the system into transfer functions that would align the LMS updates in time. Tichy also was an early investigator in acoustic intensity. By the mid-1990s, a robust frequency-domain ANC system was developed that combined both ANC and intensity to study the power flow and energy conversion in an ANC system [3]. In the early years of ANC, there was much controversy on how the additional energy from a secondary source could result in less energy radiated from a duct driven by a primary source. By using an Intensity error spectrum in a frequency domain ANC system, the power flow (active intensity) was minimized at various positions inside the duct to reveal how the active secondary source uses impedance to govern the radiated energy. This put the controversy to rest and provided clear objective evidence of how sound is actively "cancelled" or more precisely, how controlling waveguide impedance can control radiated sound power. Its an honor to have worked with Jiri.

WEDNESDAY MORNING, 9 DECEMBER 2020 — 11:15 A.M. TO 12:45 P.M.

## Session 3aAAb

## Architectural Acoustics, Noise, Signal Processing in Acoustics and Engineering Acoustics: Session in Memory of Jiri Tichy II

Victor W. Sparrow, Cochair
*Penn State, 201 Applied Science Bldg., University Park, PA 16802*

Gary W. Elko, Cochair
*mh acoustics, 25A Summit Ave., Summit, NJ 07901*

**Chair's Introduction—11:15**

### *Invited Papers*

**11:20**

**3aAAb1. Instantaneous frequency and its applications.** Hideo Suzuki (none, Sea Tower 2502, 1-2-3 Utase, Mihamaku, Chiba City, Chiba 261-0013, Japan, suzukihideo@nifty.com)

The instantaneous frequency (IF) and envelope (IE) analysis of signals is very useful in various fields. In this paper, three examples of applications are presented. (1) Representing signals by IF and/or IE. Since signals before and after allpass filtering have the same frequency spectra, there is no way to distinguish them by the spectrum analysis. However, IF and/or IE may show clear differences between them. It is interesting that IF at the onset of a sinusoidal burst signal is twice the frequency of the sinusoid (JAES, September 1980). (2) Analyzing vibratos of violin sounds recorded in CDs in order to find out differences of playing styles of virtuosi such as Grumiaux, Oistrakh, Perlman, and Stern. Some results show clear differences in the speeds and depths of the frequency modulation (vibrato) (ISMA 2004). (3) Analyzing rotation fluctuations of automobile rotating shafts. An electromagnetic or electrostatic sensor located close to a rotating gear measures a quasi-sinusoidal signal. The IF analysis of this signal gives the rotation frequency and details of its fluctuation structure. Not only the steady state rotation, but transient rotation change analysis of a reciprocating engine at the sequence of idling stop and start (ISS) are also possible (https://www.aandd.co.jp/adhome/pdf/catalog/nvh$_{analysis}$/c$_{asra}$.pdf).

**11:40**

**3aAAb2. Studies on the vibration of Japanese drum wood barrels under material property uncertainty.** Yun Fan Hwang (None, 2580 Arundel Ave. Carlsbad, CA 92009, yfhwang1@gmail.com) and Hideo Suzuki (None, Mihamaku, Chiba, Japan)

Unlike isotropic material such as steel, wood behaves as a highly orthotropic composite material. Due to measurement difficulty, only the longitudinal Young's modulus and the specific gravity of the wood were successfully measured at the time the wood barrel was made. In the experiments, mode shapes and modal resonance frequencies were measured. In the numerical modal analysis, the finite-

element model of the drum was constructed using orthotropic conical shell elements which require more elastic constants than those being measured. A try-and-error method was used to estimate those unknown constants by comparing the computed and measured modal frequencies and shapes. It was found that the estimated value of the circumferential (cross-grain) Young's modulus, which was unknown at the beginning of the study, turns out to be crucial in determining the lower mode resonance frequencies. The usefulness of this analysis is that the estimated elastic constants can be used for updating the finite-element model which can then be used for calculating the higher order modes. These higher modes cannot be practically determined by measurements. [Work by 2nd author supported by Miyamoto Unosuke Shouten Co.]

**12:00**

**3aAAb3. An overview of mixed-model inversion and its application to the study of traumatic brain injury.** Anthony J. Romano (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave., SW, Code 7160, Washington, DC 20375-0001, anthony.romano@nrl.navy.mil) and William G. Szymczak (Acoust. Div., Naval Res. Lab., Washington, DC)

Previously, we introduced Mixed-Model Inversion which utilizes a fusion of Magnetic Resonance Elastography (MRE), Diffusion Tensor Imaging (DTI), and a combination of isotropic and anisotropic inversion algorithms for the evaluation of the viscoelastic stiffness and anisotropy of the human brain. This approach was developed as a non-invasive diagnostic tool for the study of brain health or pathology. The aims of this work are (1) to evaluate the viscoelastic properties of the brains of both healthy controls and patients who present with Traumatic Brain Injury (TBI) and (2) to demonstrate the alterations to the stiffness and anisotropy of healthy brain structures as a result of insult/injury. When compared to healthy controls, there were significant differences in the stiffness as well as the anisotropic models of the white matter in the TBI patients, with alterations in dependence upon the insult/injury experienced. Preliminary studies indicate that this fusion of measurement and analytical modalities can provide metrics, based on differences in material stiffness and anisotropy, for differentiation between healthy controls and TBI patients for diagnostic purposes. [Work supported by Dr. Timothy Bentley, ONR Code 34, Warfighter Protection.]

**12:20**

**3aAAb4. Appreciations of Jiri Tichy as mentor, educator in the 1980's.** David Kahn (Acoust. Distinctions, 400 Main St., Ste. 600, Stamford, CT 06901, dkahn@ad-ny.com)

Jiri Tichy was leading Penn State's Graduate Program in Acoustics in 1980 when I applied to the program. He directed my studies there and was my advisor for my master's thesis on the topic of architectural acoustics. His mentoring continued past my graduation from the program, 3 years later until my master's thesis research topic was published in the Acoustical Society of America's journal in September 1986. Having been in the field of architectural acoustics for over 35 years since graduating his program, that perspective has helped my appreciation for what a gifted mentor and educator he was. During this presentation, I will share some of the highlights of my tutelage under Jiri Tichy.



WEDNESDAY MORNING, 9 DECEMBER 2020 — 9:30 A.M. TO 10:15 A.M.

## Session 3aAB

## Animal Bioacoustics: Animal Bioacoustics Poster Session

Authors will be at their posters from 9:30 a.m. to 10:15 a.m.

### *Contributed Papers*

**3aAB1. Recycling data: An annotated marine acoustic data set that is publicly available for use in classifier development and marine mammal research.** Kristen S. Kanes (Sci. Services, Ocean Networks Canada, 2474 Arbutus Rd., Victoria, BC V8N1V8, Canada, ksjkanes@oceannetworks.ca)

Barkley Canyon is a productive submarine canyon approximately 60 km southwest of Vancouver Island, Canada. The canyon's nutrient flow is affected by multiple regional currents, and draws aggregations of euphausiids, hake, herring, and various marine mammal species. A subset of the acoustic data collected from the 2013–2015 hydrophone deployment on the Barkley Canyon Upper Slope platform of Ocean Networks Canada's North-East Pacific Time-series Undersea Networked Experiments (NEPTUNE) observatory was manually annotated for marine mammal presence to investigate marine mammal habitat use and in support of development of a random forest classifier. This dataset, which is being made publicly available for further use, includes strong-label annotations of phonations from blue whales, fin whales, humpback whales, sperm whales, orcas, Pacific white-sided dolphins, Risso's dolphins, and other delphinids that could not be identified to species. All regional orca communities are represented within the dataset, and phonations are labelled to ecotype and pod level when possible. This dataset could be further used in a number of ways, including classifier development, investigations into habitat use and seasonality, or combining the acoustic data with data collected from co-located oceanographic instruments to investigate links between marine mammal presence and oceanographic conditions.

**3aAB2. Assembling an acoustic catalogue for different dolphin species in the Colombian Pacific coast: An opportunity to parameterize whistles before rising noise pollution levels.** Daniel Norena (Universidad de los Andes, cra 16 #127b-43, Bogotá DC 110121, Colombia, daniel.norename@gmail.com)

Growing ship traffic worldwide has led to a relatively recent increase in underwater noise, raising concerns about effects on marine mammal communication. Many populations of several dolphin species inhabit the eastern Pacific Ocean, along the coast of Colombia. Noise pollution levels in the Colombian Pacific coast (CPC) are very low. Currently, the CPC is slated for the construction of a port in the Gulf of Tribugá. Previous port construction in other countries have shown that this will change the acoustic environment and will compromise marine fauna, such as dolphins. This is the first study focused on the whistle acoustic parameters from several dolphin species in the region before any disturbance. Opportunistic recordings were made in two different locations alongside the coast, reporting five different delphinid species. The results show that the repertoire of four species is different when compared to other populations in more disturbed areas around the globe. An LDA was used to cluster the acoustic parameters and it supported the acoustic niche hypothesis, finding that these species may avoid whistle overlapping. If constructed, the port could force species to adjust their vocal repertoire engaging in an inter-specific whistle overlapping, or could lead to area abandonment, which would cause economic and ecological disasters for the region.

**3aAB3. Spectral interleaving by singing humpback whales: Signs of sonar.** Eduardo Mercado (Univ. at Buffalo, SUNY, 204 Park Hall, Buffalo, NY 14260, emiii@buffalo.edu)

Singing humpback whales produce a wide variety of tonal and broadband units within their songs, usually organized into patterned sequences. Here, an extensive database of songs made available through Google's Pattern Radio website was analyzed to test the hypothesis that singers produce consecutive units in ways that minimize overlapping spectral content. Consistent with earlier analyses, singers maintained spectral separation across units within most repeating patterns (phrases and subphrases). Additionally, singers preserved the continuity of concentrated spectral energy, as well as the timing of units, as they cycled through different themes. Within songs, singers alternated between producing units that generated persistent reverberant tails tightly focused within narrow frequency bands and more impulsive, broadband units. By spectrally interleaving units, singing humpback whales can potentially avoid overlap between units that could interfere with auditory reception of the signals. By producing signals with broadband and narrowband elements that are spectrally and temporally segregated, singers may also maximize the distances at which they can detect and localize targets using echoes from those signals.

**3aAB4. Burst-pulsed sounds of rough-toothed dolphins (*Steno bredanensis*) in Southeastern Brazil.** Isabela M. Lima (Laboratório de Mamíferos Aquáticos e Bioindicadores (MAQUA) - Faculdade de Oceanografia, Universidade do Estado do Rio de Janeiro (UERJ), Rio de Janeiro 20550-013, Brazil, isabelaseabra.lima@gmail.com), Mariana Barbosa, Tatiana Bisi, José Lailson-Brito, and Alexandre Azevedo (Laboratório de Mamíferos Aquáticos e Bioindicadores (MAQUA) - Faculdade de Oceanografia, Universidade do Estado do Rio de Janeiro (UERJ), Rio de Janeiro, Brazil)

Recordings of *Steno bredanensis* were done in Southeastern Brazilian coast during eleven days between 2013 and 2017. The system consisted of a C54-XRS hydrophone ($-165$ dB *re*: 1 V/$\mu$Pa, 0.009 to 100 kHz) and a Fostex FR-2 digital recorder with a 192 kHz sample rate. Recordings were inspected for burst-pulsed sounds in Adobe audition 1.5 software in a 512 Hann window, 50% overlap. When they were found, the sound was cut to a new file of 2-s window, and a butterworth high-pass filter was used when necessary. A burst-pulse sound was considered for analysis when its pulses were correctly and clearly identified in the Sound Ruler software, which was used to obtain the acoustic parameters (512 Hann window, 50% overlap). Thirty-one burst pulsed sounds were analyzed. The mean number of pulses per burst-pulse was $52 \pm 44$ (varying between 6 and 224), mean inter-pulse interval was $5.1 \pm 1.9$ ms (varying between 2.4 and 8.4 ms), mean burst-pulse duration was $0.27 \pm 0.23$ s (varying between 0.04 and 1.1 s), and mean peak frequency was $26.1 \pm 11.3$ kHz (varying between 7.0 and 64.7 kHz). Although preliminary, these results show a first characterization of the burst-pulsed sounds of this species.

**3aAB5. More than song and calls: New repetitive tones for humpback whales in the breeding area of Colombia.** Mar Palanca (Madre Agua Ecoturismo e Invesitgación, C/Belchite 4-13, Valencia 46009, Spain, mar.palanca.g@gmail.com), Lisa Walker (Grooved Whale Project, Vancouver, BC, Canada), Esteban Duque (Madre Agua Ecoturismo e Invesitgación, Sabaneta, Colombia), Ross Nichols (Univ. of California, Santa Cruz, CA), Kerri D. Seger (Appl. Ocean Sci., Fairfax Station, VA), Fred Sharpe (Alaska Whale Foundation, Pittsburgh, AK), Ari Friedlaender (Univ. of California, Santa Cruz, CA), and Christina E. Perazio (Univ. at Buffalo, Buffalo, NY)

The vocal repertoire of humpback whales (*Megaptera novaeangliae*) consist of long and complex songs and social calls. Songs are characterized by their cyclical and predictable structure in form of units, phrases, and themes. On the contrary, social calls are less predictable and generally short bursts of vocalizations that mainly occur during inter or intragroup interactions. During monitoring activities of reproductive stock G in Bahía Solano (Breeding ground, Colombia), a series of undescribed vocalizations were recorded, which we called repetitive tones. These are loud, frequency modulated (between 280 and 770 Hz), quickly repeated vocalizations. They are shorter than one second and have short inter-call intervals (from 0.7 to 1.0 s). They do not follow the pattern of either vocal category established in the species' repertoire. Here we discuss the occurrence of such vocalizations as a result of acoustic and behavioral plasticity of cetaceans. We explore vocal mimicry, song crystallization, or the documentation of a new single-unit song type as possible scenarios to understand the potential function(s) of these tones. Framing acoustic behavior in the cognitive and adaptive capacity of cetaceans argues for a third vocal repertoire category that could serve as the basis for contextualizing new behavioral displays.

**3aAB6. To the beat of a different drummer: Defining the rhythmic traits of humpback whale vocalizations.** Mar Palanca (Univ. of Bonn, Valencia, Spain), Lisa Walker (Grooved Whale Project, Vancouver, BC, Canada, groovedwhale@gmail.com), Esteban Duque (Madre Agua Ecoturismo e Investigación, Sabaneta, Colombia), Ross Nichols (Univ. of California Santa Cruz, Santa Cruz, CA), Fred Sharpe (Alaska Whale Foundation, Pittsburgh, AK), Kerri Seger (Scripps Inst. of Oceanogr., Santa Monica, CA), Christina E. Perazio (Univ. of Buffalo, Buffalo, NY), and Ari Friedlaender (Univ. of California Santa Cruz, Santa Cruz, CA)

Tonal signals are useful in behavioral studies as they often represent the building blocks of more complex vocalizations. Their simple structure can be indicative of fundamental processes such as resonant chamber size. Repetitive tones produced by the humpback whales from the Colombian breeding ground are not easily categorized into song nor social calls. These tones display a strong rhythmic quality, yet little is known about the functionality of temporal patterning in its vocalizations. Studies on the temporal structure of song have noted that unit-to-unit duration is relatively consistent within a phrase but that rhythm in song often lacks precision. Taking a musicological approach, we examined potential carrier signals of information within the tones by comparing its rhythmic qualities to those found in social calls (particularly feeding calls) and song, and theorize how rhythmic characteristics may aid the whale in its feeding, communication, and mating pursuits.

**3aAB7. Deteclic: A multi-method detector of sperm whale's click.** Fabio Cassiano (UMR CNRS 6285 Lab-STICC, ENSTA Bretagne, 2 rue Francois Verny, Brest 29200, France, fabio.cassiano@ensta-bretagne.org), Angélique Drémeau, Flore Samaran, and Isabelle Quidu (UMR CNRS 6285 Lab-STICC, ENSTA Bretagne, Brest, France)

In the French EEZ of Crozet and Kerguelen Islands, longline fishing is heavily impacted by the depredation of sperm whale, with significant socio-economic implication and ecological impact. Until now, only visual observations of the surface have been used to monitor this phenomenon. Their limitations have raised the need to extend the methods of investigation. Thus, an autonomous acoustic recorder was attached directly to the longline and several hours of recording were collected, disqualifying de facto manual annotation of the data. To overcome this difficulty, we developed an automatic detector, based on five simple methods from the literature: intercorrelation with a reference signal, a Teager-Kaiser energy operator convolved with a Gabor function, spectrogram analysis, kurtosis-based statistical detection and analysis of the Daubechies 15 wavelet-transform. The different methods runs independently with a click-length analysis window allowing a click-by-click detection, and then performs a vote to limit the false alarm rate. Our tool performance is assessed on a dataset where 2450 clicks have been identified by an expert. The clicks detected give valuable indication of the presence/absence of cetacean around the longline, the level of click detection allows us to dissociate the different acoustic behaviours leading to the detection of the depredation event.

**3aAB8. Humpback whale song analysis based on automatic classification performance.** Carlos A. Rueda (MBARI, 7700 Sandholdt Rd., Moss Landing, CA 95039-9644, carueda@mbari.org) and John P. Ryan (MBARI, Moss Landing, CA)

Song is a prevalent behavior in humpback whale populations, even in regions that are considered to be foraging habitat such as the Monterey Bay National Marine Sanctuary in the Northeast Pacific, where song is detected nine months out of the year. In this work, we explore various machine learning methods to classify song units, as a basis for studying song structure and its changes. We report on a number of analyses and classification exercises based on linear predictive coding, vector quantization, and machine learning classifiers including Naive Bayes, first-order Markov chain, and Hidden Markov modeling. As a baseline for comparison purposes, the distortion measure used to create the codebooks for vector quantization is itself also used as a means for classification. With classification accuracy ranging from 65% to 83% across the selected methods on a 4.5 h recording involving 5000 unit occurrences over 20 different unit types, we evaluate the effect of several signal processing, clustering, and learning parameters on classification performance with the goal of laying a foundation that can be used to characterize song vocalization at not only the unit level but also below (subunit) and above (phrase, theme, song).



WEDNESDAY MORNING, 9 DECEMBER 2020 9:30 A.M. TO 11:00 A.M.

## Session 3aBAa

## Biomedical Acoustics, Physical Acoustics, Structural Acoustics and Vibration, and Computational Acoustics: Fractional Calculus Models of Compressional and Shear Waves for Medical Ultrasound I

Sverre Holm, Chair
*Physics, University of Oslo, P. O. Box 1048, Blindern, Oslo N 0316, Norway*

**Chair's Introduction—9:30**

### *Invited Papers*

**9:35**

**3aBAa1. Finding consensus among rheological models for shear waves in soft tissues.** Thomas L. Szabo (Biomedical Eng., Boston Univ., 44 Cummington Mall, Boston, MA 02215, tlszabo@bu.edu), Kevin J. Parker (Elec. & Comput. Eng., Univ. of Rochester, Rochester, NY), and Sverre Holm (Phys., Univ. of Oslo, Oslo, Norway)

Shear wave viscoelastic properties of tissue are now measured by a wide variety of methods including elastography, imaging scanners, rheological shear viscometers, and a variety of calibrated stress-strain analyzers. Because absorption and sound speed can be strong functions of frequency, fitting the data to an viscoelastic model which best describes observed behavior is a common step in understanding and comparing data sets among tissues and disease states. Because the same data can be represented by different model constants, there is a need to reach consistency and consensus on the most effective models among different sub-fields in acoustics, biomechanics, and elastography. To this end, we examined many established rheological models as well as data sets. We argue that the long history of biomechanics, including the concept of the extended relaxation spectrum, and the theoretical framework of multiple relaxation models which model the multi-scale nature of physical tissues, consistent with power law data extending over several decades of frequency and time, all lead to the conclusion that fractional derivative models represent the most succinct and meaningful models of soft tissue viscoelastic behavior.

**9:55**

**3aBAa2. Characterization of tissue mimicking phantoms and soft tissues using ultrasound shear wave elastography and power-law models.** Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Dept. of Radiology, Mayo Clinic, Rochester, MN 55905, urban.matthew@mayo.edu), Piotr Kijanka (Dept. of Robotics and Mechatronics, AGH Univ. of Sci. and Technol., Krakow, Poland), Benjamin Wood (Dept. of Radiology, Mayo Clinic, Rochester, MN), and Robert J. McGough (Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI)

Characterization of mechanical properties of tissue has become an important area of study over the last two decades. Shear wave elastography (SWE) is being used in many clinical applications to characterize, and multiple mechanical properties are being reported. To more comprehensively characterize soft tissues, we have focused on frequency-domain methods for evaluating viscoelasticity. We measure the shear wave velocity and attenuation and the variation with frequency. We will present results with the Kelvin-Voigt fractional derivative model and general power laws to characterize data from SWE experiments in tissue-mimicking phantoms and *ex vivo* and *in vivo* tissue. Tissue mimicking phantoms from CIRS, inc., that have varying amounts of viscoelasticity will be investigated. Measurements from *ex vivo* and *in vivo* liver and kidney will also be used for this study. We will examine the range of the power-law exponents associated with these samples and the different models to provide understanding of the range of these parameters and how well tissue mimicking phantoms realize the parameters measured in soft tissues.

**10:15**

**3aBAa3. Fractional derivative group shear wave speeds for model-based viscoelastic characterization.** Courtney A. Trutna (Biomedical Eng., Duke Univ., Rm. 1427, Fitzpatrick Ctr. (FCIEMAS), 101 Sci. Dr., Durham, NC 27708, courtney.trutna@duke.edu), Ned C. Rouze, Mark L. Palmeri, and Kathryn R. Nightingale (Biomedical Eng., Duke Univ., Durham, NC)

We present a method for model-based viscoelastic characterization using shear wave elastography. The shear wave displacement signal is differentiated to a series of fractional derivative orders, and the group speed of the signal at each derivative order is calculated. These speeds are compared to a look up table built from simulations for a given viscoelastic material model. The best-fit material parameters are chosen to minimize the mean square difference between the group speeds from the look up table and experimental measurements across all fractional derivative orders. Various two- and three-parameter viscoelastic material models are tested. The technique is validated in simulations with ultrasonic tracking and demonstrated in three viscoelastic phantoms with material properties that match those associated with different degrees of liver fibrosis. Comparatively, phase velocity curves are computed, and parameters determined by fitting analytic expressions for phase velocity for a given material model. The fractional derivative group speed-based technique gives model parameters with lower mean square error than fitting material model parameters directly to phase velocities. Additionally, we conclude these viscoelastic phantoms are sufficiently characterized by a two-parameter material model and are better characterized by the Linear Attenuation model than the more commonly used Voigt model.

**10:35**

**3aBAa4. Power law dispersion from shear wave images.** Juvenal Ormachea (Elec. & Comput. Eng., Univ. of Rochester, Rochester, NY) and Kevin J. Parker (Elec. & Comput. Eng., Univ. of Rochester, Comput. Studies Bldg. 724, Box 270231, Rochester, NY 14627-0231, kevin.parker@rochester.edu)

Many normal soft tissues exhibit dispersion of shear wave phase velocity that is consistent with power law behavior. This rheology is consistent with the fractional Zener model under assumptions that appear to be realistic for normal liver and other tissues. The phenomenon can be measured using multi-frequency reverberant shear wave elastography (R-SWE). A simultaneous multi-frequency R-SWE field can be accomplished by applying an array of external sources that operate at multiple frequencies, for example, 50, 100, 150…500 Hz, all contributing to the shear wave field produced in the target organ. With estimates of phase velocity across this frequency band, the R-SWE approach can obtain a 2-D power law coefficient (PLC) image. The clinical feasibility of this method has been analyzed by assessing the shear wave speed (SWS) and PLC in phantoms and *in vivo* human organs. For *in vivo* liver cases, mean SWS (1.99 m/s, 2.29 m/s, and 2.38 m/s) and mean PLC (0.23, 0.44, and 0.43) were estimated for a thin patient, and for two obese patients, respectively. For these cases, the PLC results may be an additional parameter that could help to differentiate the viscoelastic properties in liver tissue and enhanced image contrast in cases where lesions or pathologies show an altered viscoelastic response compared with normal tissue.

## Session 3aBAb

## Biomedical Acoustics: General Biomedical Acoustics: Tissue Engineering

Jonathan A. Kopechek, Cochair
*Bioengineering, University of Louisville, 2301 S. Third St., Paul C. Lutz Hall, Rm. 419, Louisville, KY 40292-0001*

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

**Chair's Introduction—9:30**

### *Contributed Papers*

**9:35**

**3aBAb1. Time-dependent, microscale mechanical properties of acoustically responsive scaffolds using atomic force microscopy and confocal imaging.** Mitra Aliabouzar (Univ. of Michigan, 3218-02 Med Sci I, 1301 Cathreine St., Ann Arbor, MI 48109, aliabouz@med.umich.edu), Christopher D. Davidson, William Y. Wang, Oliver D. Kripfgans, J.Brian Fowlkes, Brendon M. Baker, and Mario L. Fabiilli (Univ. of Michigan, Ann Arbor, MI)

Acoustic droplet vaporization (ADV) has been used to modulate delivery of regenerative molecules from acoustically responsive scaffolds (ARSs), which are composite fibrin hydrogels containing payload-carrying, phase-shift emulsions. Here, we studied the micromechanical properties of ARSs including stiffness as well as elastic and shear moduli over 7 days post-ADV. ARSs, containing dextran-loaded perfluorohexane (PFH) emulsions (diameter: 6 mm) embedded in fluorescently labeled fibrin gels, were exposed to acoustic pressures above the ADV threshold ($2.2 \pm 0.2$ MPa ) at 2.5 MHz. The compressive stiffness as well as elastic modulus of fibrin gels (i.e., without emulsions) and ARSs (pre- and post-ADV) were measured via force-spectroscopy and the Hertz model, respectively. The measured stiffness was lowest ($1.9 \pm 0.5$ mN/m) in regions adjacent to the PFH emulsions and highest ($16.0 \pm 4.4$ mN/m) in regions adjacent to the ADV-generated bubbles. Fibrin compaction at the bubble–fibrin interface was observed using time-lapsed confocal imaging and correlated with bubble growth at the time points studied here. In addition, different ADV-generated bubble responses in ARSs will be discussed using time-lapsed confocal imaging. Elucidating the mechanical microenvironment within the ARS could be used to control mechanically induced, cellular processes and further the understanding of ADV-triggered drug delivery for regeneration.

**9:55**

**3aBAb2. Acoustic and mechanical characterization of gelatin methacryloyl scaffolds for tissue engineering applications.** Megan Anderson (George Washington Univ., 800 22nd St. NW, Washington, DC 20052, andersonm@gwmail.gwu.edu), Jenna Osborn (Mech. and Aerosp. Eng., George Washington Univ., Washington, DC), Raj Rao, Lijie Grace Zhang, and Kausik Sarkar (George Washington Univ., Washington, DC)

Gelatin methacryloyl (GelMA) is a highly biocompatible, biodegradable material and a 3-D printable option for constructing tissue engineering scaffolds. Improved material characterization of GelMA is necessary to optimize preparation techniques, evaluate tissue similarities, and validate tissue engineering potential. Conventional testing methods are through destructive, one-time measurements, yet nondestructive tests are desired for evaluating long-term changes. In the present study, ultrasound techniques were utilized to measure mechanical properties of GelMA tissue scaffolds. Varying concentrations of GelMA and ultraviolet light curing time produced scaffolds with a range of material properties. Ultrasound pulse-echo techniques were used for a non-destructive acoustic characterization procedure, and parameters including speed of sound, acoustic impedance, and attenuation coefficient were measured. To further evaluate the material properties of the scaffolds, compression testing was performed. Physical parameters of GelMA were found to be similar to those of native tissues, demonstrating that GelMA scaffolds are biomimetic. The impact of GelMA concentration and curing time will be discussed to inform the selection of preparation parameters for specific tissues. Acoustic characterization proves to be a promising technique for evaluating the structure and function of the scaffolds and could serve as an indicator of tissue scaffold health while providing real-time monitoring *in vivo*.

**10:15**

**3aBAb3. Non-thermal effects of ultrasound are selectively induced during a critical window of collagen self-assembly.** Emma G. Norris (Pharmacology and Physiol., Univ. of Rochester, 601 Elmwood Ave., Box 711, Rochester, NY 14642, emma_grygotis@urmc.rochester.edu), Joseph B. Majeski, Sarah E. Wayson, Holly Coleman, Regine Choe, Diane Dalecki (Dept. of Biomedical Eng., Univ. of Rochester, Rochester, NY), and Denise C. Hocking (Pharmacology and Physiol., Univ. of Rochester, Rochester, NY)

Type I collagen is a self-assembling fibrillar protein widely used in tissue engineering. An established paradigm utilizes ultrasound exposure during the fluid-to-gel phase transition of neutralized collagen solutions, producing site-specific changes in collagen fiber organization via both thermal and non-thermal acoustic mechanisms. In the present study, we investigated the temporal dependence of ultrasound bioeffects with respect to the progression of collagen assembly. Collagen solutions (0.8 mg/ml) were exposed to ultrasound (CW, 7.8 or 8.8 MHz, 0–10 W/cm$^2$) for 5 min during three distinct stages of collagen self-assembly, which were monitored via both temperature and optical turbidity measurements. Assembly rate was manipulated by adjusting the collagen pH and the temperature at which acoustic exposures were performed. The results identify a critical window during which ultrasound exposure produces radially aligned collagen fibers via a non-thermal acoustic mechanism. This window coincides with a stage of collagen polymerization during which nanofibrils associate laterally into microfibril bundles, and a simultaneous increase in the acoustic absorption coefficient. These findings raise the possibility that ultrasound exposures of self-assembling collagen biomaterials in a point-of-care clinical setting can be timed to preferentially induce either thermal or non-thermal bioeffects, thereby enhancing the efficacy of therapeutic ultrasound for regenerative medicine applications.

10:35

**3aBAb4. Feasibility of a single-transducer harmonic motion imaging using frequency-based simultaneous multiple harmonic oscillation excitation pulses.** Md. Murad Hossain (Biomedical Eng., Columbia Univ., 630 West 168 St., P&S 19-418, New York, NY 10032, mh4051@columbia.edu) and Elisa Konofagou (Biomedical Eng., Columbia Univ., New York, NY)

Single-transducer harmonic motion imaging (ST-HMI) uses a single transducer to generate the harmonic motion at a particular frequency by modulating excitation pulse duration and then, estimate the motion by collecting tracking pulses in-between the excitation pulses, unlike the conventional HMI which uses focused ultrasound and an imaging transducer. Instead of oscillating at a single frequency, the objective is to use ST-HMI for simultaneously exciting the tissue at multiple frequencies. Six excitation pulses per period were selected by sampling a continuous signal which was generated by summing sinusoids with fundamental and harmonics of 100 Hz and 200–1000 Hz, respectively. A mean peak to peak displacement (MP2PD) image was generated by a weighted average of P2PD at each frequency with a weighting factor derived from the Fourier transform. The new method is evaluated by imaging four inclusions (INC, Young's modulus: 6, 9, 32, and 75 kPa) embedded in 18 kPa background (BKD) and HIFU ablated excised canine liver. The MP2PD image successfully differentiated all four inclusions with $R^2 = 0.99$ of the linear regression between the MP2PD ratio of BKD to INC versus Young's modulus ratio of INC to BKD. The MP2PD image detected the ablated region with the MP2PD ratio of non-ablated versus ablated regions of 1.64. These results demonstrate the feasibility of simultaneously exciting the tissue at multiple frequencies. Ongoing studies entail the translation of this new elasticity imaging method of detecting breast masses in humans.

WEDNESDAY MORNING, 9 DECEMBER 2020 — 11:15 A.M. TO 12:45 P.M.

## Session 3aBAc

## Biomedical Acoustics, Physical Acoustics, Structural Acoustics and Vibration, and Computational Acoustics: Fractional Calculus Models of Compressional and Shear Waves for Medical Ultrasound II

Sverre Holm, Chair
*Physics, University of Oslo, P. O. Box 1048, Blindern, Oslo N 0316, Norway*

**Chair's Introduction—11:15**

### *Invited Papers*

11:20

**3aBAc1. Numerical evaluation of a fractional calculus model for the spatial impulse response of a circular piston.** Drew A. Murray (Comput. Sci. and Eng., Michigan State Univ., Michigan State University, East Lansing, MI 48824, murraydr@msu.edu) and Robert J. McGough (Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI)

In lossless media, exact analytical expressions for the spatial impulse response describe the effect of diffraction in the time domain for several specific transducer shapes. At present, no exact analytical expressions are available for lossy media, so numerical evaluation of the spatial impulse response is required. For attenuation that follows a power law, the spatial impulse response is numerically evaluated by superposing contributions from time-domain Green's functions for the Power Law Wave Equation, which is a fractional calculus model for power law attenuation. To demonstrate examples of lossy spatial impulse responses obtained with these time-domain Green's functions, which are expressed in terms of maximally skewed stable densities, numerical results are computed for a circular transducer using different attenuation values and compared with the analytical result for a lossless medium. The results show that the numerically computed spatial impulse response for a lossy medium converges to the analytical result evaluated in a lossless medium as the value of the attenuation constant decreases. As the attenuation constant grows larger, the temporal extent increases, and the sharp edges are replaced by increasingly smooth curves, as observed in numerical evaluations of the lossy spatial impulse response evaluated in multiple locations.

11:40

**3aBAc2. Application of fractional calculus to the ultrasonic characterization of human bone tissue.** Zine El Abiddine Fellah (Mech. and Acoust. Lab., CNRS LMA, Lab. of Mech. and Acoust., 4 impasse Nikola Tesla CS 40006, 13453 Marseille Cedex 13, France, Fellah@lma.cnrs-mrs.fr), Remi Roncen (ONERA, Toulouse, France), Mohamed Fellah (Physique Théorique, USTHB, Algiers, Algeria), Erick Ogam (LMA, CNRS, Marseille, France), and Claude Depollier (LAUM, Le mans, France)

Transient ultrasonic propagation in human bone tissue is considered according to Biot's theory. The bone is modeled as a porous medium with an elastic structure. The viscous fluid/structure interactions are described by fractional calculus in the time domain. The slow and fast compressional waves, as well as the rotational shear wave predicted by Biot's theory obey fractional propagation equations. The fractional equation system is solved analytically in the time domain, thus obtaining the expressions of the reflection and transmission scattering operators. Inverse identification of the intrinsic microstructure of the pores and of the mechanical properties of the bone is performed in the time domain (waveforms) and frequency domain (attenuation and phase velocities), by adopting a statistical Bayesian inference technique using ultrasonic transmitted and reflected signals, which allows to find the identified parameters and their associated uncertainty.

12:00

**3aBAc3. On the origin of frequency power-law for tissue mechanics in elastography.** Ralph Sinkus (INSERM UMRS1148— Lab. for Vascular Translational Sci., INSERM, GH Bichat-Claude Bernard— 46 rue Henri Huchard, Paris 75877, France, ralph.sinkus@inserm.fr), Giacomo Annio, Gregory Franck (INSERM UMRS1148— Lab. for Vascular Translational Sci., INSERM, Paris, France), and Sverre Holm (Phys., Univ. of Oslo, Oslo, Norway)

The imaginary part of the complex shear modulus in tissue is not negligible. In liver the phase angle (ranging between 0 and 1) is about 0.2 while in kidney it is about 0.3. The presence of dispersion can have its origin either in a constitutive loss—i.e., absorption of energy—or in scattering of the wave and hence represents an apparent loss. Since dispersion in tissue follows over a wide range a frequency power-law, fractional order derivative models such as the springpot model are well suited to fit the data in the clinically accessible range of 30–200Hz. They, however, do not provide a fundamental understanding of whether loss is due to friction and hence conversion to heat, or due to material heterogeneities and thus scattering. Models such as ODA are able to relate the observable frequency power-law to the spatial distribution of scatterers. If loss in low-frequency elastography were solely due to scattering, this would render the method extremely powerful in characterizing for instance blood vessel architecture in oncology. To distinguish whether the measured prominant loss in porcine tissue (phase angle $\sim$0.4) is originating from absorption or scattering, we use MR-Spectroscopy to measure absolute temperature via the resonance-frequency-shift between water and methylene, at precisions better than 0.1 °C. Temperature should increase theoretically by about 1/2 °C at an amplitude of 10 $\mu$m, 500 Hz, and an exposure of 1000 s, which is not observed. This points towards scattering as the main mechanism for wave attenuation.

12:20

**3aBAc4. Justification for power laws and fractional models.** Sverre Holm (Phys., Univ. of Oslo, P. O. Box 1048, Blindern, Oslo N 0316, Norway, sverre.holm@fys.uio.no)

Wave equations with non-integer derivative operators describe attenuation which increases with frequency with other powers than two, unlike ordinary wave equations. It is desirable to try to understand what properties of, e.g., biological tissue that give rise to this behavior. The main attenuation mechanisms of standard acoustics are heat conduction and relaxation, as well as structural and chemical relaxation. They have fractional parallels and the first one is heat relaxation described by fractional Newton cooling due to anomalous diffusion. The most important mechanism is however the fractional parallel to structural relaxation. Instead of one there are multiple relaxation processes with a distribution of relaxation times that follows a power-law distribution, possibly indicating fractal properties. The distribution also has a relationship to the Arrhenius equation, indicating a link to chemical relaxation, albeit a quite speculative one. The multiple relaxation model may also be formulated as a hierarchical polymer model. Time-varying non-Newtonian viscosity and a medium with a fractal distribution of scatterers can also give rise to power law behavior. Existing models in sediment acoustics such as the grain shearing model and the Biot poroelastic model can also be reformulated with fractional operators. These approaches are presented in the hope of progressing towards an understanding of whether fractional wave equations give clues to some deeper reality, or if they are just a compact phenomenological description.

# Session 3aBAd

## Biomedical Acoustics: General Biomedical Acoustics: Quantitative Ultrasound

Jonathan A. Kopechek, Cochair
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Xiaoming Zhang, Cochair
*Mayo Clinic, 200 1st St. SW, Rochester, MN 55905*

**Chair's Introduction—11:15**

### *Contributed Papers*

**11:20**

**3aBAd1. The link between internal microbubbles in kidney stones and the Doppler twinkling artifact.** Eric Rokni (The Penn State Univ., State College, PA 16801, ezr144@psu.edu), Scott Zinck, and Julianna Simon (The Penn State Univ., State College, PA)

The color Doppler ultrasound twinkling artifact, which appears as a rapid color shift on some kidney stones, has recently been attributed to stabilized microbubbles. However, it is unknown what stone features allow these bubbles to form. Here, thirteen kidney stones were mechanically scanned in water using a research ultrasound system (Vantage, Verasonics, Kirkland, WA) at 2.5 MHz (C5-2, Philips, Bothell, WA), 5 MHz (L7-4, Philips, Bothell, WA), and 18.5 MHz (L22-14v, Verasonics, Kirkland, WA). Three-dimensional maps of Doppler power were generated using saved I/Q data. Then, six stones were imaged with either environmental scanning electron microscopy (ESEM) (Philips/FEI XL-20) or underwater micro-computed tomography ($\mu$CT) (GE v|tome|x L300, Wunstorf, Germany) at ambient and hypobaric pressures. Twinkling on all 13 stones was consistent over repeated trials and significantly lower at 18.5 MHz compared to 2.5 and 5 MHz. ESEM showed water condensed on the smallest surface pores ($\sim$1 $\mu$m diameter) as humidity increased. $\mu$CT showed gas trapped inside 2/3 stones that may contribute to twinkling. Future work includes investigating other pathological mineralizations to determine the role of chemical composition and growing conditions on bubble formation and twinkling. [Work supported in part by the National Science Foundation 1943937.]

**11:40**

**3aBAd2. Towards real-time muscle fatigue and recovery markers using a wearable continuous wave Doppler ultrasound system.** Joseph Majdi (George Mason Univ., 4400 University Dr., Fairfax, VA 22030, jmajdi2@masonlive.gmu.edu), Parag V. Chitnis, and Siddhartha Sikdar (George Mason Univ., Fairfax, VA)

Functional electrical stimulation (FES) is commonly used in physical rehabilitation, bypassing the central nervous system to activate motor neurons directly. However, the unnatural muscle recruitment pattern induced by FES causes rapid muscle fatigue, greatly reducing the muscle's ability to generate force. Currently there exists no reliable, real time indicator for FES-induced muscle fatigue. We believe that signs of muscle fatigue can be inferred from medical ultrasound. Previously we investigated tissue Doppler imaging (TDI) to study muscle physiology associated with muscle potentiation and fatigue. Here, we expand on that research using continuous wave (CW) Doppler ultrasound to create a wearable, low power muscle fatigue monitor. We are investigating this system to work with a hybrid FES exoskeleton designed to use the patient's own muscles with FES with the added stability of an exoskeleton. CW ultrasound indicated that the duration of muscle recruitment decreased from 129.0 ms to 51.7 ms for the same FES as the muscle fatigued. Further, we showed that muscle twitch duration and velocity correlate with twitch force, a marker of fatigue recovery, using TDI and CW. These fatigue and recovery measures can be used to inform the exoskeleton controller to coordinate FES and electric motors for producing gait.

**12:00**

**3aBAd3. Influence of stickiness on structure function of biology cells.** Quang Tran (Univ. of Illinois at Urbana-Champaign, 306 N. Wright St., Urbana, IL 61801, qntran2@illinois.edu), William D. O'Brien, and Aiguo Han (Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Structure function (SF) is critical in improving the accuracy of quantitative ultrasound (QUS) and tissue characterization. Existing SF models in the field of QUS tissue characterization are based on a hard-sphere assumption where the inter-scatterer potential consists solely of a repulsive hard core. This paper investigates a sticky hard-sphere (SHS) model containing two major characteristics of a real potential, a repulsive potential and an attractive potential. The SHS model assumes two spheres tend to be pulled closer when they are very close to each other. This model can be relevant to biological cells as they are adhesive. This paper presents a theoretical SF for the SHS model with monodisperse scatterer size. To explore the scatterer spatial distribution, a simulation of the SHS model for low, medium, and high volume fractions (0.10, 0.32, and 0.64) at different degrees of the unitless stickiness (0.2, 0.6, and 1.7) over the frequency range from 11 to 105 MHz is introduced. A comparison between the theoretical and simulated SFs shows that the two SFs are in good agreement, indicating the theoretical SF describes the scatterer distribution successfully. This study emphasizes the importance and influence of the stickiness parameter for the SF. [Work supported by R01CA226528.]

## Session 3aCAa

# Computational Acoustics, Underwater Acoustics, Structural Acoustics and Vibration, and Acoustical Oceanography: Acoustic Optimization: Methods and Applications I

Micah Shepherd, Chair
*The Pennsylvania State University, PO Box 30, State College, PA 16801*

**Chair's Introduction—9:30**

### *Contributed Paper*

**9:35**

**3aCAa1. Modeling of electromagnetically excited noise and vibration of induction motor.** Xiaorui Hu (School of Automotive Studies, Tongji Univ., No. 4800, Cao'an Rd., Jiading District, Shanghai 201804, China, hubrook@163.com) and Shuguang Zuo (School of Automotive Studies, Tongji Univ., Shanghai, China)

For electric vehicles, acoustic noise of traction motor excited by electromagnetic force is one of the main sources of noise, especially when automobiles are driven at low speed where noise from other sources are comparatively low. To analyze the characteristics of the noise and propose noise reduction suggestions, an accurate noise prediction model is needed. In this paper, a multi-physics model to predict the electromagnetic force excited acoustic noise of induction motor is presented. First, a three-dimensional (3-D) transient electromagnetic model of the motor was established using finite-element method (FEM). The uneven distributed time-varying magnetic force was calculated, and the current harmonics due to pulse width modulation (PWM) supply were considered. Then, a structural model was built, in which the anisotropy characteristic of silicon sheets stacked stator core, as well as the influence of windings, were considered through the assignment of orthotropic parameters to the stator core. Through mesh mapping, the FEM calculated magnetic force was transferred onto the structural model and the forced vibration was calculated using mode superposition method. Base on that, the acoustic noise was further obtained using boundary-element method (BEM). Finally, the noise and vibration of the studied induction motor were tested in a semi-anechoic room. As the simulated results show a good accordance to the measured data, the accuracy of the model was verified.



### *Invited Papers*

**9:55**

**3aCAa2. Large-scale gradient-based optimization in acoustics applications.** Timothy Walsh (Simulation Modeling Sci., Sandia National Labs., MS 0897, Albuquerque, NM 47906, tfwalsh@sandia.gov)

Engineering applications in acoustics and structural acoustics commonly require optimization with a large, high-dimensional space of parameters. Examples include inverse problems, design, and control, especially in cases where the parameters to be optimized are spatially and/or temporally dependent. In these scenarios the dimension of the search space precludes the use of global-based optimization algorithms, due to the rapidly growing number of required numerical evaluations of the partial differential equation (PDE) (e.g., wave equation). In such cases, gradient-based optimization with adjoint methods provides an attractive strategy wherein the cost of evaluating sensitivities is independent of the number of optimizable parameters. In this talk, we will present an overview of adjoint-based PDE-constrained optimization in Sandia's Sierra Mechanics software, with a focus on acoustic and structural acoustics applications. We will derive a general mathematical framework and then specialize to acoustics applications. Examples will be given in inverse, design, and control that involve spatially or temporally dependent variables with high-dimensionality. *[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 No. DE-NA0003525.]*

**10:15**

**3aCAa3. On acoustic prediction models for the intrinsic parameters of rigid-frame porous media using Bayesian optimization.** Kirill V. Horoshenkov (Dept. of Mech. Eng., Univ. of Sheffield, Sheffield S10 2TN, United Kingdom, k.horoshenkov@sheffield.ac.uk), Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY), and Alistair Hurrell (Dept. of Mech. Eng., Univ. of Sheffield, Sheffield, United Kingdom)

Knowledge of several intrinsic material parameters is crucial to the modeling accuracy of sound propagation in porous media. This paper studies the performance of several popular acoustic prediction models used to invert key intrinsic parameters of rigid frame porous media such as characteristic lengths, permeabilities and median pore size. These prediction models along with acoustic surface impedance experimentally measured with a standard impedance tube setup are evaluated in an optimization process which is essentially a

Bayesian estimator. The effect of the number of intrinsic parameters in the model, sample thickness and frequency range on the accuracy of the Bayesian estimator is studied. The quality of model predictions is characterized using quantitative measures such as mean, variance and the interdependence of the estimated parameters. This paper also discusses uncertainties of the intrinsic parameters and effect of these uncertainties on the accuracy of the acoustic parameter prediction.

**10:35**

**3aCAa4. Shape optimization of a compression driver phase plug taking visco-thermal losses into account.** Martin Berggren (Dept. of Computing Sci., Umeå Univ., Campustorget 5, Umeå 90187, Sweden, martin.berggren@cs.umu.se), Anders Bernland (Dept. of Computing Sci., Umeå Univ., Umeå, Sweden), André Massing (Dept. of Mathematics and Mathematical Statistics, Umeå Univ., Umeå, Sweden), Daniel Noreland (Skogforsk, Umeå, Sweden), and Eddie Wadbro (Dept. of Computing Sci., Umeå Univ., Umeå, Sweden)

The compression driver, used to feed midrange horns, consists of a compression chamber whose outlets are connected to the horn throat through a *phase plug*. The main challenge in the design of the phase plug is to avoid resonance and interference phenomena. The complexity of these phenomena makes it difficult to accomplish this design task manually. Therefore, we employ an algorithmic technique that combines numerical solutions of the governing equations with a gradient-based optimization algorithm that almost arbitrarily can deform the walls of the phase plug. A particular modeling challenge here is that visco–thermal losses cannot be ignored, due to the presence of narrow chambers and slits in the driver. Fortunately, a recent accurate but computationally inexpensive boundary-layer model is applicable and is here successfully used within the optimization loop. We use this model together with the so-called Cut Finite Element technique to avoid mesh changes when the geometry is modified by the optimization algorithm. Applying these techniques, the algorithm was able to successfully design the shape of a set of radially-directed phase plugs so that the final frequency response closely matches an ideal response, that is, one that is obtained by a lumped circuit model, ignoring wave effects.

WEDNESDAY MORNING, 9 DECEMBER 2020 — 11:15 A.M. TO 12:45 P.M.

## Session 3aCAb

## Computational Acoustics, Underwater Acoustics, Structural Acoustics and Vibration, and Acoustical Oceanography: Acoustic Optimization: Methods and Applications II

Micah Shepherd, Chair
*The Pennsylvania State University, PO Box 30, State College, PA 16801*

**Chair's Introduction—11:15**

### *Invited Papers*

**11:20**

**3aCAb1. A comparison of algorithms for the vibroacoustic optimization of a beam: Gradient-based versus evolutionary.** Cameron A. McCormick (The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, cam634@psu.edu) and Micah Shepherd (The Penn State Univ., State College, PA)

Two common classes of optimization algorithm used in structural and/or multidisciplinary optimization are gradient-based algorithms and evolutionary algorithms. In the case of smooth, unimodal objective spaces, gradient-based algorithms are generally faster, requiring fewer iterations to reach a solution. Evolutionary algorithms, on the other hand, are more robust against objective spaces that are nonlinear, discontinuous, and multimodal. This talk will present an optimization of the thickness distribution of a cantilever beam, inspired by a similar study carried out by Berggren *et al.* ["Sound vibration damping optimization with application to the design of speakerphone casings," in 10th World Congress on Structural and Multidisciplinary Optimization (2013)]. The objective is to minimize the vibration response within a certain region of the beam at discrete frequencies, with constraints on total mass and static compliance. The objective space is expected to be nonlinear and potentially multimodal. A transfer matrix method is used to evaluate the objective function and constraints, and optimal solutions are found using both a gradient-based algorithm and an evolutionary algorithm. Qualitative and quantitative results will be presented in comparing the optimized distribution to that of Berggren *et al.* and in discussing the benefits and limitations of the two algorithms for vibroacoustic optimization.

11:40

**3aCAb2. Quantitative metrics to experimentally benchmark geoacoustic inversion methods.** Julien Bonnel (Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS# 11, Woods Hole, MA 02543-1050, jbonnel@whoi.edu)

Geoacoustic inversion schemes are complex methods that combine various experimental designs (e.g., arrays or single receivers, fixed or moving sources,…), different data processing methods (e.g., tracking ray arrivals, filtering modes, estimating spectral matrices,…) and different inversion methods (e.g., brute force grid search, stochastic optimization,…). Because of this complexity and variability, geoacoustic inversion methods are difficult to benchmark. Although simulated benchmarks are possible, experimental benchmarks are more difficult since (1) experimental ground truth is never fully available and (2) different methods may sense different features/scales of the seabed. Nonetheless, assessing the performance of geoacoustic inversion methods is of paramount importance, since the resulting geoacoustic models are used to predict acoustic field for various applications, including Navy operations or noise pollution monitoring. In this presentation, we propose quantitative metrics to assess the experimental performances of inversion methods. The proposed metrics allow inter-comparison of various geoacoustic models, but also comparison of predicted acoustic field with experimental (independently collected) data. The metrics are successfully applied on a subset of data collected during the Shallow Water 2006 experiment.

12:00

**3aCAb3. Broadband acoustic lens design using the principle of reciprocity and gradient-based optimization.** Feruza Amirkulova (Mech. Eng., San Jose State Univ., 1 Washington Sq, San Jose, CA 95192, feruza.amirkulova@sjsu.edu), Samer Gerges (BASIS Independent Silicon Valley, San Jose, CA), and Andrew N. Norris (MAE, Rutgers, Piscataway, NJ)

In this work, we will present a gradient-based optimization method for the design of an acoustic lens. This design is based on an optimization process using the semi-analytical optimization approach and applying the principle of reciprocity. Here, we will illustrate the use of acoustic reciprocity to define the pressure at the focal point due to a source located in a far-field. The reciprocity enables us to relate the response by configuration of scatterers for an incident plane wave in terms of the far-field Green's function. The idea differs from earlier inverse designs that use topology optimization tools and generic algorithms. The gradient-based optimization algorithm maximizes the sound amplification at the focal point by evaluating pressure derivative with respect to the cylinder positions and then perturbatively optimizing the position of each cylinder in the lens while taking into account acoustic the multiple scattering between the cylinders. Computations are performed on MATLAB using "fmincon" function with a MultiStart optimization solver, and supplying the gradient of pressure at the focus. The reciprocal formulation of inverse design of acoustic lens leads to the closed-form solution that takes lesser time to compute the absolute pressure and it gradients at the focal point. The numerical results for the optimization of the broadband acoustic lens will be illustrated for configurations of uniform rigid cylinders.

## *Contributed Paper*

12:20

**3aCAb4. Optimization approach for designing diffuse acoustic fields on demand.** Wilkins Aquino (Mech. Eng. and Mater. Sci., Duke Univ., Hudson Hall, Durham, NC 27708, wa20@duke.edu) and Jerry Rouse (Sandia National Labs., Albuquerque, NM)

Reverberation chambers are used for design, optimization and qualification of structures. The low frequency limit (Schroeder frequency) of applicability is dependent upon the chamber size. We have developed a source optimization methodology which can decrease this cut-on frequency. The target diffuse field is represented by plane waves having uniformly distributed direction and phase. The corresponding cross-spectral density can be shown to be a Bessel function of the first kind in 2-D and a sinc function in 3-D. The construction of the diffuse field at a given frequency (or set of frequencies) is then cast as a constrained optimization problem. To this end, the goal is to find the cross-spectral density of a set of point sources that closely approximates (in some sense) the target diffuse field cross-spectral density (e.g., sinc or Bessel function). To enforce positive definiteness of the solution, we use a matrix factorization approach in which the matrix factors become the design variables. We demonstrate that we can construct diffuse fields on demand in arbitrary enclosures with known wall impedance. Furthermore, we demonstrate that there exist source configurations that can generate near diffuse fields at frequencies below the Schroeder frequency.

# Session 3aEAa

## Engineering Acoustics and Architectural Acoustics: Microphones: From Rock Stars to Rockets I

Vahid Naderyan, Cochair
*University of Mississippi, 1151 Maplewood Dr., Itasca, IL 60143*

Sandra J. Guzman, Cochair
*Shure, Inc., 4800 W Touhy Ave., Niles, IL 60714*

Edward M. Okorn, Cochair
*GRAS NA Inc., 2234 East Enterprise Parkway, Twinsburg, OH 44087*

Neil A. Shaw, Cochair
*Menlo Scientific, P O Box 1610, Topanga, CA 90290*

**Chair's Introduction—9:30**

### *Invited Papers*

**9:35**

**3aEAa1. The microphone engineering lab notebooks of Ben Bauer.** Michael S. Pettersen (Corporate Historian, Shure Inc., 5800 W. Touhy Ave., Niles, IL 60714, pettersen_michael@shure.com)

Benjamin B. Bauer (1913–1979) was an ASA Fellow and awarded the ASA Silver Medal (1978). He held over 100 patents for acoustical/audio technology, with his first patent being arguably the most significant: the invention of the Uniphase principle integral to the Shure Unidyne model 55 microphone. Introduced in 1939 and still manufactured today, the Shure Unidyne was the first unidirectional microphone using a single dynamic element. Today, the Uniphase principle is employed in the vast majority of directional microphones. In September 2016, Bauer's microphone engineering lab notebooks, dating from 1936 to 1944, were found in offsite storage; they had not been seen for over 50 years. The presentation provides a peek into these Bauer notebooks as he discovers and refines the Uniphase principle, as well as numerous other electro-acoustical concepts—some decades ahead of their time.

**9:55**

**3aEAa2. Microphones that predate modern electroacoustics.** Stephen C. Thompson (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, sct12@psu.edu)

Our modern understanding of Electroacoustics includes acoustic sensors (microphones) and projectors (loudspeakers) and the electronic amplifiers that form a complete electroacoustic system. The first electronic amplifiers were made with vacuum tubes and became available about a century ago. Thus, we may consider electroacoustics as a technology whose history began at that time. The first microphones were invented before then, but most microphone technologies needed electronic amplification for practical employment. On the other hand, the telephone communication industry, which began in the 1870s, was reasonably mature by the 1920s using a design that did not include vacuum tubes. This presentation will describe the devices used in the early telephone system and how that system could exist without electronic amplification.

**10:15**

**3aEAa3. Off-axis frequency response in a dual diaphragm, uniphase moving coil microphone.** Roger S. Grinnip (Acoust. Eng., Shure Inc., 5800 W. Touhy Ave., Niles, IL 60714, grinnip_roger@shure.com)

When the rear port in a uniphase, unidirectional condenser microphone is modified to include a series compliance element (dual diaphragm) the low frequency performance can be improved. The benefit is most notable in the off-axis response as it relates to source/receiver proximity (proximity effect). Due to the significant operational differences between uniphase moving coil and condenser microphones, incorporating a dual diaphragm element into the rear port of a moving coil microphone requires careful attention to the design. Simplified network models of the uniphase system, proximity effect, and dual diaphragm moving coil design are presented. Measured results of a dual diaphragm, uniphase moving coil microphone are presented with off-axis response comparisons to a single diaphragm, uniphase moving coil microphone.

**10:35**

**3aEAa4. Avoiding the audio range: Frequencies very high and very low.** Thomas Gabrielson (Penn State Univ., PO Box 30, State College, PA 16804, tbg3@psu.edu)

Condenser measurement microphones have been an essential component of acoustic measurements for many years. Although used most often in the "audio range" (20 to 20 000 Hz), they are also capable of quality measurements both well above and well below that range if their limitations are respected and if their outputs are processed carefully. For example, below 20 Hz, the natural low-frequency roll-off provides an automatic pre-whitening for ambient spectra, thereby reducing the dynamic range required, and the roll-off can be compensated by a simple pole-moving filter in post-processing. Well above 20 kHz, diffraction can be significant, the characteristics of an analog-to-digital converter's anti-aliasing filter can be important, and transit time across the surface of the microphone's membrane may not be negligible. Nowhere are these characteristics more important than in measurement of the rise time of acoustic shock waves. If accurate reproduction of the time-domain waveform is important, then any compensation schemes must account for the system phase response as well as the magnitude response. A diffraction-compensation filter can be effective in minimizing waveform over-shoot and a carefully designed baffle can delay the onset of diffraction effects if leading-edge phenomena like peak pressure and rise time are important.

WEDNESDAY MORNING, 9 DECEMBER 2020 11:15 A.M. TO 12:25 P.M.

## Session 3aEAb

## Engineering Acoustics and Architectural Acoustics: Microphones: From Rock Stars to Rockets II

Vahid Naderyan, Cochair
*University of Mississippi, 1151 Maplewood Dr., Itasca, IL 60143*

Sandra J. Guzman, Cochair
*Shure, Inc., 4800 W Touhy Ave., Niles, IL 60714*

Edward M. Okorn, Cochair
*GRAS NA, Inc., 2234 East Enterprise Parkway, Twinsburg, OH 44087*

Neil A. Shaw, Cochair
*Menlo Scientific, PO Box 1610, Topanga, CA 90290*

**Chair's Introduction—11:15**

### *Invited Papers*

**11:20**

**3aEAb1. Novel functional microphones created using 3D printing for rapid development.** James Windmill (Univ. of Strathclyde, University of Strathclyde, 204 George St., Glasgow G1 1XW, United Kingdom, james.windmill@strath.ac.uk) and Andrew Reid (Univ. of Strathclyde, Glasgow, United Kingdom)

The ability to detect sound is a sense found across multiple animals in the natural world. Hearing in the insects has, for example, evolved independently approximately nineteen times. This has produced a huge variety of different miniature acoustic sensory structures, with many different characteristics and novel features. Many different researchers have attempted to take inspiration from biological hearing systems, with the most well-known being the miniature directional ear of the *Ormia ochracea*. However, from an engineering point of view there are specific problems involved. Firstly, the acoustic structures found in the natural world are three dimensional, and composed of softer biological materials. Secondly, the development cycle for researchers working on MEMS based microphones depends on their ability to access fabrication facilities. Whilst the use of multi-user fabricators reduces costs, particularly for academic research, the turnaround time is measured in months. This talk will present recent work to utilise digital light processing 3-D printing techniques as a method to rapidly create novel microphone designs. The initial development of acoustic devices that utilise 3-D structures will be illustrated, and the use of multiple materials in single 3-D prints to create functional microphones capable of producing an electrical output signal will be discussed.

**11:40**

**3aEAb2. Design of capacitive acoustic sensors using highly compliant microbeam arrays.** Mahdi Farahikia (Mech. Eng., SUNY Binghamton, Binghamton, NY) and Ronald Miles (Mech. Eng. Dept., SUNY Binghamton, 85 Murray Hill Rd., Binghamton, NY 13902, miles@binghamton.edu)

We have previously shown that the viscous forces acting on a periodic array of infinitesimally thin micro-beams due to air-borne sound are frequency-independent and directional [Mahdi Farahikia and Ronald Miles, "Viscous flow sensing using micro-beam arrays," *J. Acoust. Soc. Am.* **146**(4), 2837–2837 (2019)]. Utilizing this phenomenon in the design of acoustic sensors requires a means of converting their structural motion into electronic signals. Capacitive sensing, as a method for this purpose, requires the combination of both fixed and moving electrodes while eliminating the instability associated with high bias voltages [Ronald N. Miles, "A compliant capacitive sensor for acoustics: Avoiding electrostatic forces at high bias voltages," *IEEE Sens. J.* **18**(14), 5691–5698 (2018)]. In this study, the motion of moving micro-beams adjacent to fixed micro-breams in periodic arrays due to air-borne sound is examined through the finite element method. It is concluded that thinner, narrower micro-beams lead to a desired flat frequency response. The optimum gap between the micro-beams is found to be between 0.5 and 1 times the micro-beam width. The results from this study are fundamental in implementing a capacitive sensing mechanism to fabricate miniature acoustic sensors using micro-beam arrays.

**12:00**

**3aEAb3. Evolution of miniature microphones in hearing instruments.** Janice L. LoPresti (Hearing Health Technologies, Knowles Corp., 1151 Maplewood Dr., Itasca, IL 60143, janice.lopresti@knowles.com)

The invention of the transistor enabled early hearing instruments to be miniaturized: from bulky body worn device to a device that was more visually appealing and could fit in the ear canal. This change also brought many technical challenges related to the miniaturization of the microphone components and creation of low power amplifiers to maintain battery life. In the early 1950s, magnetic microphones were considered a good choice with their smaller size and lower output impedance compatible with the transistor amplifiers of that time period. However a susceptibility to shock and vibration drove further innovation leading to first piezoelectric and then electret microphone designs. The next large step in the design evolution occurred in the early 2000s with the introduction of the MEMS (Micro-Electro-Mechanical Systems) microphone. The adoption of MEMS microphones in hearing industry is driven by multiple factors including environmental stability requirements for beam forming applications and reflow compatibility for manufacturing. Modern technology drivers such as sound quality, miniaturization, reliability, and advanced audio controls have pushed the limits of current technology to create best performing microphones today. This presentation chronicles evolution of miniature microphones in hearing instruments from first magnetic microphones to MEMS microphones in present day.

WEDNESDAY MORNING, 9 DECEMBER 2020 — 9:30 A.M. TO 11:00 A.M.

## Session 3aEDa

## Education in Acoustics, Physical Acoustics, Structural Acoustics and Vibration, Noise, and Musical Acoustics: Acoustics Demonstrations for Classroom Teaching I

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

**Chair's Introduction—9:30**

### *Invited Papers*

**9:35**

**3aEDa1. Acoustics outreach demonstration show: Sounds to astound.** Cameron T. Vongsawad (Dept. of Phys. and Astronomy, Brigham Young Univ., 820 W 555 N, Orem, UT 84057, cvongsawad@byu.edu), Aaron B. Vaughn, Carla Wallace, Tracianne B. Neilsen, Brian E. Anderson, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The sounds to astound acoustics demonstration show is put on twice a year for the local community by the Brigham Young University Student Chapter of the Acoustical Society of America. The show is a free, interactive demonstration show that explores the science of sound for a target audience of 5th to 8th grade students. A variety of introductory acoustics concepts such as waves, frequency, amplitude, and medium are taught in conjunction with an associated demonstration, animation, and/or video. This presentation will describe

the planning and execution of the demonstration shows as well as identify several demonstrations and the corresponding acoustics concept(s) taught with it. In addition to these shows, tours of the anechoic and reverberation chambers are also provided for appropriate age groups.

**9:55**

**3aEDa2. Propellers, motors, and fans.** Thomas Gabrielson (Penn State Univ., PO Box 30, State College, PA 16804, tbg3@psu.edu)

Effective demonstrations make immediate visual and aural impact but also inspire discussion, analysis—even argument. From quick study to in-depth examination, inexpensive drone motors and small cooling fans provide excellent objects for classroom demonstration of the acoustics of rotating machinery. One of the most fundamental concepts—blade passage rate—is straightforward to show, to understand, and to relate to the acoustic signature. From there, the process of discovery expands to shaft rotation rate, blade thickness and loading noise, blade/strut interaction, flow noise, gear-mesh noise, electrical-to-mechanical coupling through magnetic-field interaction, and the impact of pulse-width modulation of the drive signal on the acoustic radiation. Processing can be as simple as visualization of a microphone output using an oscilloscope or a spectrogram display; however, the processing can grow to synchronous averaging and rotation-rate compensation with the addition of straightforward optical or electrical measurements. A relatively simple apparatus provides a broad spectrum of concepts in a highly visible form.

**10:15**

**3aEDa3. Acoustic-phonetics demonstrations for classroom teaching.** Takayuki Arai (Information and Commun. Sci., Sophia Univ., 7-1 Kioi-cho, Chiyoda-ku, Tokyo 102-8554, Japan, arai@sophia.ac.jp)

Physical demonstrations using vocal-tract models have been shown to be a quite intuitive way to teach acoustic phenomena on speech production for acoustic phonetics and speech science classes. Several models for different purposes have been developed by Arai within the last 20 + years, including vocal-tract models, sound sources, and lung models [e.g., T. Arai, *J. Acoust. Soc. Am.* **131**(3), 2444–2454 (2012)]. By combining them, the following topics can be effectively taught in virtual and in-person classrooms: visualizing the sound propagation, the relationship between vocal-tract configuration and sound quality; the source-filter theory of speech production; the quantal theory; many-to-one mapping between articulation and sound; many-to-one mapping between sounds and phoneme; and source-filter interactions. Because such demonstrations are highly demanded, the Acoustic-Phonetics Demonstrations (APD) website has been made public (www.splab.net/APD/) to make videos and sound demonstrations available for academic use. The same website contains files of subsets of the vocal-tract models for 3D-printers, so users can download and 3D-print them to have their own models. In addition, some of the models are being used for workshops and exhibitions at museums. The applications of the models are now expanding to multiple areas, such as language learning and clinical situations like speech therapy.



**10:35**

**3aEDa4. Demonstration and validation of nonlinear acoustic shock wave formation in a renovated waveguide.** Connor J. McCluskey (Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, cxm1198@psu.edu), Mark A. Langhirt, Nick E. Carder, and Luke A. Wade (Penn State Univ., University Park, PA)

When acoustic waves attain high enough pressure amplitudes, the wave speed becomes amplitude-dependent, causing the wave to steepen. If the amplitude is high enough, a shock is formed. Shock formation is commonly covered in nonlinear acoustics courses, but it is difficult to demonstrate experimentally in a classroom setting. Dr. Isadore Rudnick designed a nonlinear waveguide demonstration, as recalled by Dr. Robert Keolian [*J. Acoust. Soc. Am.* **145**(3 Pt. 2), 1682 (2019)]. In his setup, a high amplitude driver inputs an acoustic plane wave into a long PVC pipe. At the Pennsylvania State University, a similar waveguide was built by Dr. Lauren Falco as part of her research on nonlinear jet noise. The waveguide was then used as part of an assignment in a laboratory class taught by Dr. Steven Garrett. However, the waveguide has not been used for almost 10 years. This demonstration apparatus has been restored to working order and was validated by measuring the nonlinear generation and decay of harmonics at three locations along the pipe by varying the peak pressure and frequency of the sinusoidal input. This paper will demonstrate the formation of shock waves using this apparatus.

# Session 3aEDb

## Education in Acoustics, Physical Acoustics, Structural Acoustics and Vibration, Noise, and Musical Acoustics: Acoustics Demonstrations for Classroom Teaching II

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

**Chair's Introduction—11:15**

### *Invited Papers*

**11:20**

**3aEDb1. Photoacoustic imaging demonstration.** Leah E. Burge (Phys., U.S. Naval Acad., Annapolis, MD 21402, leahburge02@gmail.com) and Murray S. Korman (Phys., U.S. Naval Acad., Annapolis, MD)

Experiments [T. J. Allen and P. C. Beard, "High power visible light emitting diodes as pulsed excitation sources for biomedical photoacoustics," *Biomed. Opt. Express* **7**(4), 1260–1270 (2016)] describe the use of a visible light-emitting diode (LED) as an alternative to using a Q-switched Nd:YAG laser as a photoacoustic excitation (1 mJ pulse) source for below-the-surface vascular imaging. They describe the feasibility of using a low-cost LED, overdriven by a factor of 10 in current, with short pulses $\sim$200 ns operating at a 0.01% duty cycle to achieve $\sim$10 $\mu$J in an imaging experiment involving hemoglobin absorption (wavelength of 623 nm). A version of the experiment is demonstrated where an SST-90R LED ($\lambda = 620$ nm, 18 A maximum CW current) is driven from a home-made MOSFET pulsed driver, capable of providing pulses up to 50 A. Our demonstration of photoacoustic tomography involves a $\sim$1cm diam LED beam illuminating three closely-spaced vertical 1.4 mm tubes filled with blood (35% haematocrit), mounted in an open acrylic water tank. An immersion transducer (3.5 MHz, cylindrically focused PZT with 60 dB gain) detects time-averaged photoacoustic signals versus rotation angle. Then back-projection imaging can construct a two-dimensional image.

**11:40**

**3aEDb2. Demonstration of subjective sound preferences and jury studies in product noise control using countertop blenders.** Andrew R. Barnard (Mech. Eng., Michigan Technol. Univ., 815 R.L. Smith MEEM Bldg., 1400 Townsend Dr., Houghton, MI 49931, arbarnar@mtu.edu), Troy Bouman, and Suraj Prabhu (Mech. Eng., Michigan Technol. Univ., Houghton, MI)

Developing consumer products requires consideration of product noise. Product noise control is usually not as simple as measuring and reducing the sound pressure level as much as possible. For example, consumers will not purchase vacuum cleaners if they are too quiet because of a perception that they do not function as well as louder models. This demonstration uses three countertop blenders to teach students about subjective acoustics and jury studies. The demonstration will start with a blind acoustic comparison of three blenders and the audience will be asked for opinions on the blenders, based on sound alone. Those opinions will be compared to overall measured sound pressure levels, product costs, and product performance criterion. This demonstration is ideal for an introductory noise control course, or for community outreach as it is easily relatable for any level of audience, from K–12 to adults.

**12:00**

**3aEDb3. An acoustic resonator rocket: Reviving Dvorak's 1878 acoustic repulsion apparatus.** Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, dar119@psu.edu)

When a Helmholtz resonator is driven with sufficiently high amplitude, the pressure difference between the inside and outside of the resonator and the non-uniform oscillating flow of air into and out of the resonator opening results in a reaction force on the resonator akin to rocket thrust. The earliest demonstration of "acoustic repulsion" using small Helmholtz resonators was reported by M. Dvorak [*Philos. Mag. Ser. 5* **6**(36), 225 (1878)]. Lord Rayleigh explained that the average pressure inside the resonator is higher than the external atmospheric pressure, with the result that the "resonator tends to move as if impelled by a force" [*Philos. Mag. Ser. 5* **6**(37), 270 (1878)]. This demonstration will show an updated version of Dvorak's 1878 apparatus using Christmas ornament balls as suggested by Uno Ingard, [*Notes on Acoustics* (Infinity Science Press, 2008), p. 336]. In addition to showing the resonator rocket in action and discussing its history and various theorical explanations, this demonstration will show how the amount of "thrust" may be measured with a mass balance and will attempt to visualize the flow in the neck of the oscillator.

# Session 3aMU

## Musical Acoustics: General Topics in Musical Acoustics I

Bozena Kostek, Cochair
*Audio Acoustics Lab., Gdansk Univ. of Technology, Narutowicza 11/12, Gdansk, 80-233, Poland*

Tim Ziemer, Cochair
*Bremen Spatial Cognition Center, University of Bremen, Enrique-Schmidt-Str. 5, Bremen, 28359, Germany*

**Chair's Introduction—11:15**

### *Contributed Papers*

**11:20**

**3aMU1. Acoustics of the French Horn.** Natalie West (Phys. Dept., Loyola Univ. Chicago, Loyola University Chicago, Chicago, IL 60660, nwest2@luc.edu) and Gordon P. Ramsey (Phys., Loyola Univ. Chicago, Chicago, IL)

This study explored the French horn and its various properties that contribute to its unique timbre. With an emphasis on the role of the right hand, the variables studied were: type of material, manufacturer, right hand position with respect to the bell, the range of notes, and player. The project mainly included experiments at Loyola University Chicago and was supported using Northwestern University's anechoic chamber. Horns are cylindrical-conical hybrids and are able to produce many harmonics, creating a rich sound. The right hand in the bell can be used for quick intonation adjustments and its placement has large effects on the timbre. Our analysis determined the variable that most contributed to the unique sound was the right hand position, while the type of horn (a combination of material and manufacturer) and the range of notes played also had moderate effects on timbre. We will present the theory surrounding the horn's shape and various components, experimental procedure, and analysis of our results. The significance of the results to the horn player will be discussed.

**11:40**

**3aMU2. Two methods for acoustic modeling of the saxophone mouthpiece.** Song Wang (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol. (CIRMMT), Schulich School of Music, McGill Univ., 555 Rue Sherbrooke Ouest, Montreal, QC H3A 1E3, Canada, song.wang5@mail.mcgill.ca), Gary Scavone, and Esteban Maestre (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol. (CIRMMT), Schulich School of Music, McGill Univ., Montreal, QC, Canada)

The mouthpiece is one of the key components in determining the sound and the playability of the saxophone. Assuming plane-wave propagation inside the mouthpiece, two different models are proposed for acoustic modeling of the saxophone mouthpiece. First, a Transfer Matrix (TM) model is derived from a finite element model and is then validated by comparing the calculated input impedance of a mouthpiece-cone structure with measurements. Then, by adapting a methodology traditionally used in modeling the human vocal tract, a Geometric Model (GM) is used to approximate the mouthpiece as a series of acoustic tubes of varying cross-sectional areas, and its accuracy is examined by comparing it with the TM mouthpiece model. We couple both models to a measured input impedance of an alto saxophone and provide a comparison of both methods and their impact on different fingering configurations. Finally, we discuss applications in sound synthesis and in mouthpiece design.

**12:00**

**3aMU3. The soprano saxophone adapted to different tone generators in comparison to prototype instruments.** Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., School of Architecture, 110 8th St., Troy, NY 12180, braasj@rpi.edu)

The purpose of this project was to investigate the extent to which a prototype wind-instrument resonator could be equipped with various tone generators to meet basic expectations of Western music practice in terms of range, tonal balance, and intonation. Sound generators from a Renaissance cornett, a bassoon, a Chinese bawu, and an Armenian duduk were adapted to a soprano saxophone body to test this. Further, the saxophone was performed as a rim flute by blowing against the neck, and an extended didjeridu adapter was crafted as well. The adaptations were tested against the original instruments by comparing single tone recordings throughout the range of the instruments. The test criteria were the breadth of tonal range, achievable intonation accuracy, dynamic range, and the constancy of the frequency spectrum. Some of the variations, like the bawu-reed adaptation, were easy to play; others, like the cornett, required years of practice. It was possible to play each adapted instrument in tune throughout the range of the prototypical instrument from which the tone-generator was taken. The tonal spectra were generally between those of the original instruments, which provided the tone generator, and the spectrum of the soprano saxophone, which lent the body.

**12:20**

**3aMU4. Finite element simulation of reed-resonator coupling: The khaen pipe as an example.** Brian R. Hassard (Dept. of Phys. and Astronomy, Univ. of Utah, Salt Lake City, UT 84112, brian.r.hassard@gmail.com) and James P. Cottingham (Phys., Coe College, Cedar Rapids, IA)

Recently some work has been done on the coupling of a free reed to a resonator using finite element modeling software. This has been focused primarily on the pipes of the khaen, a Southeast Asian free-reed bamboo pipe instrument. The broad goal is realistic modeling of an air-driven reed by simulating fluid-structure interactions in conjunction with air flow and pressure fluctuations. In the current study, coupling between the reed and the resonator has been explored and compared for cylindrical cross-section pipes, including khaen pipes, as well as for Helmholtz resonators. The effects of pressure and fluid flow on the reed motion were studied, and reed-resonator coupling explored for both the Helmholtz resonator and khaen systems. Reed vibration and sound radiation have been analyzed and compared to published models. Preliminary results are realistic, including the sound radiation pattern from a reed driven khaen pipe. Some were unexpected, including a relatively large role of the second transverse mode of the reed tongue. Some unresolved discrepancies remain between results for laminar flow models and pressure physics models, which are the subject of ongoing work. Future goals include modeling a reed-driven simulation of the complete khaen instrument system.

# Session 3aNSa

## Noise, Architectural Acoustics, and Structural Acoustics and Vibration: In Memory of Richard Lyon I

Gregory C. Tocci, Cochair
*Cavanaugh Tocci Associates, Inc., 327F Boston Post Rd., Sudbury, MA 01776*

Patricia Davies, Cochair
*Ray W. Herrick Laboratories, Purdue University, 177 S Russell St., West Lafayette, IN 47907-2099*

**Chair's Introduction—9:30**

### *Invited Papers*

**9:35**

**3aNSa1. Richard Lyon's early years on the faculty of MIT's Department of Mechanical Engineering.** Allan D. Pierce (Retired, PO Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537, allanpierce@verizon.net)

Richard Lyon joined the faculty in 1970, but was associated with the ME Department for several years. Stephen Crandall, the Head of the Applied Mechanics Division, had instituted various graduate courses, including one taught by Lyon (BBN) and another taught by Miguel Junger (CAA). Allan Pierce joined as an Assistant Professor in 1966 and was tasked with teaching a graduate course on wave propagation. Lyon had emerged as the principal exponent of statistical energy analysis (SEA), and several MIT students did theses supervised or co-supervised by Lyon. Dick came as a professor in 1970. At the same time he founded Cambridge Collaborative with Jerry Manning, a fellow BBN employee and a former MIT student. Lyon, after joining the faculty, instituted several new research programs, loosely classified as architectural acoustics and noise control. The present talk focuses in part on Lyon's influence on Allan's career during the time they overlapped at MIT and gives some perspective on Lyon's style and leadership. Dick's work on acoustical scale-modeling and the theses he supervised during his early years on the faculty were of great interest to Allan. Dick prompted the work with Wayne Kinney on the field experiments in South Boston on the propagation of noise from low-flying helicopters into urban canyons. He also prompted Allan into attacking the problem of propagation around a three-sided barrier, and it was Dick who palmed off his job as associate editor of JASA onto Allan and thereby cemented his commitment to the ASA.

**9:55**

**3aNSa2. Statistical phase analysis and Lyon statistical mode shape functions.** Richard DeJong (Calvin Univ., 1712 Knollcrest Circle, Grand Rapids, MI 49546, dejong@calvin.edu)

While many of us were still trying to figure out the magnitude of SEA predictions, Dick Lyon had moved on to determining the statistics of the phase of frequency response functions. He derived a new set of functions which he named Beranek functions. This was an important element of his work on machinery diagnostics. This presentation will review the development of Statistical Phase Analysis and some following work leading to the definition of Lyon Statistical Mode Shape Functions.

**10:15**

**3aNSa3. Phase spectral characteristics of room transfer function and coherent length in reverberation sound field.** Mikio Tohyama (Wave Sci. Study, 2-7-11, Fujigaoka, Kugenuma, Fujisawa-shi, Kanagawa 251-0031, Japan, mikio.tohyama@gmail.com) and Yoshinori Takahashi (Dept. of Information Design, Kogakuin Univ., Tokyo, Japan)

The authors had studied multi-degree-of-freedom vibration system and room transfer function in a reverberant sound field focusing on zeros in the transfer function and its phase spectral characteristics, under the leadership of Professor R. H. Lyon for 20 years since 1988. Since 2006, room acoustics and wave propagation in rooms have been investigated with an inspiration from zero-drift model (the movement of zeros on the complex frequency plane) on the sound field close to a source, which was one of the most important Lyon's hypotheses. In this report, the behavior in the distribution of minimum-phase zeros is visualized by means of the narrow-frequency-band analysis for phase-frequency-characteristics on the minimum-phase component of impulse responses measured in a reverberation room. Additionally, it demonstrates that the inclination of the linear regression line for the phase-frequency-characteristics indicates "propagation phase" conjectured by Lyon with the use of the zero-drift model, namely, "phase change" proportional to the distance between a sound source and a sound receiving point. Furthermore, it shows that the range where the propagation phase is observed in the phase-frequency-characteristics agrees with the coherent length where sound waves propagate as spherical waves, which was inferred by P. M. Morse and R. H. Bolt (1944).

**10:35**

**3aNSa4. Richard Lyon and the statistics of vibration.** James McDaniel (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, jgm@bu.edu), Alyssa Liem, and Allison Kaminski (Mech. Eng., Boston Univ., Boston, MA)

Richard Lyon's pioneering work on the statistics of vibration produced significant insights and tools that profoundly improved our world. This work and its impact will be reviewed. This lecture begins with his 1955 Doctoral Dissertation at the Massachusetts Institute of Technology, which was entitled The Response of Continuous Systems to Random Noise Fields. It was supervised by K. Uno Ingard. His work on Statistical Energy Analysis followed and received widespread interest, resulting in commercial software packages that implemented the theory. For this and related work he received the Gold Medal from the Acoustical Society in 2003, "… for sustained leadership and extensive contributions in the application of statistical concepts to structural acoustics and noise." To illustrate the impact of his work and to honor his legacy, this presentation reviews recent work by the authors on the statistics of vibrations analyzed by machine learning algorithms. A training set is constructed by computing the responses of a system with randomly generated properties from known distributions. A machine learning algorithm uses this training set to predict the response of a system with known properties.

WEDNESDAY MORNING, 9 DECEMBER 2020 — 11:15 A.M. TO 12:25 P.M.

## Session 3aNSb

## Noise, Architectural Acoustics, and Structural Acoustics and Vibration: In Memory of Richard Lyon II

Gregory C. Tocci, Cochair
*Cavanaugh Tocci Associates, In, 327F Boston Post Road, Sudbury, MA 01776*

Patricia Davies, Cochair
*Ray W. Herrick Laboratories, Purdue University, 177 S Russell Street, West Lafayette, IN 47907-2099*

**Chair's Introduction—11:15**



### *Invited Papers*

**11:20**

**3aNSb1. Richard H. Lyon: Sound propagation, acoustic modeling, and the MIT A&V Lab.** Paul Donavan (Illingworth & Rodkin, Inc., 429 E Cotati Ave., Cotati, CA 94931, pdonavan@illingworthrodkin.com)

In the mid 1970s, the Acoustics and Vibration Laboratory at the Massachusetts Institute of Technology was an exciting place for any student interested in acoustics and noise control. As one of the directors of the lab, Professor Lyon had a profound influence on the activities, students, and researchers working and studying there. Among the other things that Dick is known for such as Statistical Energy Analysis, Machinery Noise Diagnosis, and Product Sound Quality, he was also overseeing research on urban sound propagation and applications of acoustical scale modeling. This presentation will discuss some of this research, the culture of the A&V Lab, and personnel interactions with Professor Lyon from one of his student's and researcher's point of view.

**11:40**

**3aNSb2. Of knitting needles and surround sound.** Christopher Blair (Akustiks, LLC, 93 North Main St., Norwalk, CT 06854, cblair@akustiks.com)

The author began his association with Dick Lyon in 1972 when he became a graduate student in the Acoustics and Vibration Lab at MIT. Their personal and professional relationship continued, on and off, through the mid 1990s. While early studies of outdoor sound propagation and transportation noise will be discussed, this presentation will focus primarily on their work together at RHLyonCorp in the study of vibration-induced failure mechanisms and in commercial product development, including early efforts in sound quality jury trials.

**12:00**

**3aNSb3. Dick Lyon's contributions to product sound quality.** David L. Bowen (Acentech, 33 Moulton St., Cambridge, MA 02138, dbowen@acentech.com)

Dick was a firm believer in bringing context of use into any problem involving how to improve the "sound quality" of a product or, as he liked to say, how to make products "sound good." He first got interested in this field when a sewing machine company approached him at M.I.T. and said they wanted their machine to sound more like their competitor's. Dick took up this challenge in his own way, saying we'll make it better sounding. He did this by incorporating various statistical techniques used in the food and flavor industries to optimize particular aspects of consumer response, substituting the sound levels of different components and mechanisms in the sewing machine for the amounts of different flavorings used in a food product. By exposing a consumer listening panel made up of actual product users to the sounds of a range of "virtual" products made up by combining the original and altered sounds of the different components (but never altered beyond what was physically feasible), and asking panelists to rate these sounds in terms of, say, acceptability, a mathematical model could be formed between changes to the component sounds and consumer ratings, thus providing concrete guidance for improved response. This and other sound quality techniques he pioneered will be discussed.

WEDNESDAY MORNING, 9 DECEMBER 2020 9:30 A.M. TO 11:00 A.M.

## Session 3aPAa

## Physical Acoustics: General Topics: Potpourri II

Curtis Rasmussen, Chair
*University of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78712*

**Chair's Introduction—9:30**

### *Contributed Papers*

**9:35**

**3aPAa1. Nonlinear resonance of sphere-plane contacts inside a cylindrical container.** Kevin Y. Lin (Phys. and Astronomy, Univ. of , 145 Hill Dr., PO Box 1848, Oxford, MS 38677-1848, klin@go.olemiss.edu), Wayne Prather (National Ctr. for Physical Acoust., Oxford, MS), and Joel Mobley (Phys. and Astronomy, Univ. of MS, University, MS)

Cylindrical containers, such as liquid tanks, pressure vessels or nuclear dry storage casks, are ubiquitous in storage applications. In our previous work, we developed metrics to estimate the structural integrity of the mock-up fuel assemblies inside a simplified lab-scale nuclear dry storage cask using sensors located strictly on the outside surface. However, the linear acoustic techniques applied have limited ability to differentiate between an absent mock-up fuel assembly and one that has fully decayed. In this work, we further examine this problem by studying contact nonlinearity in a more simplified system using Nonlinear Acoustic Resonance Spectroscopy. This system consists of a single layer of spheres of different composition and size evenly distributed at the bottom of an open aluminum cylindrical container. No extra normal force other than gravity was exerted on the spheres. The excitation source and the sensor are located at the outside surface. The system was driven close to the (1, 2) global bending mode at various excitation amplitudes, and the resonance frequencies as a function of the amplitude were recorded. The results show that this type of contact nonlinearity is mostly affected by the total mass of the spheres inside, while the diameter and composition of the spheres play minor roles. A phenomenological model was developed based on the experimental results and investigated using Finite Element simulations.

**9:55**

**3aPAa2. Application of dual-stage exhaust system using an expansion chamber and resonator for formula SAE.** Yuval Philipson (Mech. Eng., The Cooper Union for the Advancement of Sci. and Art, 9 Orchard St., New York, NY 10003, philipso@cooper.edu) and Martin S. Lawless (Mech. Eng., The Cooper Union for the Advancement of Sci. and Art, New York, NY)

Typical vehicle mufflers utilize a series of expansion chambers and resonators to reduce engine noise. Both components behave like acoustic filters by changing the impedance of the exhaust system. Expansion chambers act as low-pass filters, which mitigate noise above a cut-off frequency determined by the geometry of the chamber. Meanwhile, resonators target specific frequencies to attenuate like a notch-filter. The present work explored tuning a muffler system containing an expansion chamber and a resonator to reduce engine noise of a Formula SAE racecar. The exhaust system design was completed using custom MATLAB scripts, Ricardo WAVE, and iterative prototyping. After determining the frequencies where the engine produced maximum sound output, the components were fabricated to damp the amplitude of the offending frequencies. The transmission loss of the prototype exhaust system was tested in an anechoic chamber by measuring the frequency responses of each separate component and the entire system. Based on this experimentation, the exhaust system had an expected overall transmission loss of 45 dB. Installed on the vehicle, the operational transmission loss was 18 dB. This discrepancy is likely due to operational conditions that could not be replicated in the laboratory experiment, such as increased temperature and gas flow.

**10:15**

**3aPAa3. Spherical reflection for a planar boundary: Inverse Laplace transform of Fresnel's reflection coefficient.** Charles Thompson (Elec. and Comput. Eng, UMASS Lowell, 1 Univ. Ave., Lowell, MA 01854, charles_thompson@uml.edu), Kavitha Chandra, Pratik Gandhi, Ambika Bhatta (Elec. and Comput. Eng, UMASS Lowell, Lowell, MA), and Miroslava Raspopovic (Comput. Sci., Metropolitan Univ., Belgrade, Serbia)

Numerous investigators have examined the problem of spherical wave reflection from the planar interface between two media. The Sommerfeld's integral can be referred to in this context. Recently the inverse Laplace transform was used by Lindell and Alanen, for the case of an electric dipole above a ground plane, to improve and simplify the analysis of this problem. The current work will focus on the Laplace transform method and the required inversion of the Fresnel reflection coefficient. It is shown that the pole-zero cancellation present the reflection coefficient can result in a significant simplification in the inversion process. Application to the analysis of multilayer media is discussed.

**10:35**

**3aPAa4. The performance of time reversal in elastic chaotic cavities as a function of volume and geometric shape of the cavity.** Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT 84602, bea@byu.edu) and Paige E. Barker (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

Time reversal is used as an energy-focusing technique in nondestructive evaluation applications. Here, it is often of interest to evaluate small samples or samples that do not lend themselves to the bonding of transducers to their surfaces. A reverberant cavity, called a chaotic cavity, attached to the sample of interest provides space for the attachment of transducers as well as an added reverberant environment. Reverberation is critical to the quality of time reversal focusing. The goal of this research is to explore the dependence of the quality of the time reversal focusing on the size and geometric shape of the chaotic cavity used. An optimal chaotic cavity will produce the largest focusing amplitude, best spatial resolution, and linear focusing of the time reversed signal. Ultrasonic, elastic-wave experiments are performed on rectangular, cylindrical, and 3-D Sinai billiard prism samples. Experiments are repeated each time these samples are successively cut down to smaller volumes. As the size of the cavity decreases, the peak amplitude may increase or decrease depending on the normalization scheme employed. The higher the degree of ergodicity of the cavity, the higher quality focusing achieved.

WEDNESDAY MORNING, 9 DECEMBER 2020 9:30 A.M. TO 11:00 A.M.



## Session 3aPAb

## Physical Acoustics: General Topics: Turbulent Media

Samuel P. Wallen, Chair

*Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758*

**Chair's Introduction—9:30**

### *Contributed Papers*

**9:35**

**3aPAb1. Simulation of *N*-wave propagation in turbulent atmosphere using standard and wide-angle parabolic equations.** Petr V. Yuldashev (Faculty of Phys., M.V. Lomonosov Moscow State Univ., Leninskie Gory, Moscow 119991, Russian Federation, petr@acs366.phys.msu.ru), Maria M. Karzova, Vera Khokhlova (Faculty of Phys., M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), and Philippe Blanc-Benon (LMFA UMR CNRS 5509, Ecole Central de Lyon, Ecully, France)

Interest to the problem of sonic boom propagation in atmosphere is growing in recent decade due to the development of new generation of supersonic business jets. Important nonlinear wave effects occur in the last kilometer above the ground where sonic boom wave interacts with turbulence of the planetary boundary layer. Focusing and defocusing of the sonic boom on random inhomogeneities of sound speed lead to fluctuations of its peak overpressure and rise time, affecting the perceived loudness. Various one-way model equations of different complexity have been developed and actively used by the research community for analyzing wave propagation in turbulent atmosphere. The basic equation is the nonlinear parabolic equation of the Khokhlov-Zabolotskaya-Kuznetsov-type, which has limitation on diffraction angles. Improvement in accuracy of the diffraction term can be reached using wide-angle formulation. Here, the simulation results for N-wave propagation through homogeneous isotropic turbulence are compared for standard and wide-angle parabolic equations of scalar type. Turbulence wind velocity fluctuations are accounted via effective sound speed. The wide-angle model is based on split-step Pade approximation of the propagation operator. The importance of taking into account large diffraction angles for *N*-wave propagation in atmospheric turbulence is discussed. [Work supported by RSF-18-72-00196 and ANR-10-LABX-0060/ANR-16-IDEX-0005].

**9:55**

**3aPAb2. Analytical sound pressure expression of vortices near a ground surface.** Ambika Bhatta (Elec. and Comput. Eng., Physical and Life Sci. Solutions LLC, UMass Lowell, Lowell, MA 01854, ambikabhatta@gmail.com)

In this work, an analytical expression is sought for the sound pressure of aircraft wake vortices near a ground surface. The analytical basis for the mechanisms of aircraft vortex sound generation follows the vortex dynamic

models and vortex sound theory. The theory focuses on acoustic signatures of the vortex created from the subsonic flow. Vortex acoustic signature from the ground can mainly be characterized based on the boundary condition and transition of the flow moving away from the ground. Of interest is the acoustic characterization of wake vortex in ground effect. To this end, the method of images was used to simulate the presence of the ground. The method of images assures that the vorticity is mirrored about the ground plane that contributes to the modified amplitude above the ground. The extension of the presented approach to model the pair of vortices, impedance boundary, and turbulent flow in the ground effect will also be highlighted.

**10:15**

**3aPAb3. Characteristic spectral analysis of turbulent-flow-induced noise in rivers.** James R. Brady (Phys. and Nuclear Eng., West Point, 309 C Alexander Pl, West Point, NY 10996, james.brady@westpoint.edu), Kent L. Gee, and Mark K. Transtrum (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

This talk describes the characteristic spectra resulting from turbulent water flow in multiple sections of the Provo and Colorado Rivers in Utah. For reference these rivers are characterized as medium-sized bedrock rivers, with Strahler orders of 5 and 7, respectively. Larson Davis 831C sound level meters were used to obtain two-second, one-third octave time histories. The spectra appear to be the superposition of two characteristic spectral curves possibly resulting from two different interactions within the turbulent section of the river. This two-source model for characterizing turbulent flow induced acoustics is similar to that observed in both the characteristic spectra for jet aircraft and for turbulent plasma flow. The higher frequency curve peaks between 700 and 1100 Hz and is almost certainly due to the Minnaert Frequency associated with resonating bubbles in the stream. This process and spectral curve appear similar to the fine spectral curve associated with Jet aircraft noise. The second curve is lower in frequency and peaks around 150 Hz and is most likely a result of interaction between the turbulent water and the ground.

**10:35**

**3aPAb4. Analysis of acoustic radiation from instability waves within an off-design supersonic jet flow using combined theory and large-eddy simulation.** Jianhui Cheng (Mech. and Aerosp. Eng., Univ. of Florida, 939 Ctr. Dr., Gainesville, FL 32611, chengjianhui@ufl.edu), James D Goldschmidt, Weiqi Shen, Ukeiley Lawrence, and Steven A. Miller (Mech. and Aerosp. Eng., Univ. of Florida, Gainesville, FL)

The impact of large-scale turbulent structures on the aerodynamic flowfield and far-field radiated noise is investigated through analysis of an overexpanded supersonic jet. The jet operating conditions are $M_j = 1.3$ and Reynolds number $1.6 \times 10^6$. The Kirchhoff surface (KS) method is used to predict radiated noise based on large-eddy simulation (LES) and applied to calibrate amplitudes of an instability wave model. The acoustic pressure time history in the near- and far-field are constructed with the instability wave model. Predictions are validated with experiment and compare favorably. Cross-correlation and cross-spectral analysis shows that the noise from instability waves is highly correlated the upstream near-field and downstream far-field radiation directions. However, the radiated instability noise in the upstream direction is dominated by fine-scale noise. Results show that it is possible to design a control system for large-scale structure noise based on upstream control if the instability noise can be extracted.

## Session 3aPAc

# Physical Acoustics: General Topics: Jet Noise and Aeroacoustic Measurements

Kevin M. Leete, Chair
*BYU Department of Physics and Astronomy, Brigham Young University, N283 ESC, Provo, UT 84602*

**Chair's Introduction—11:15**

## *Contributed Papers*

**11:20**

**3aPAc1. Near to far field relationship of crackle-related events in military aircraft jet noise.** Aaron B. Vaughn (Dept. of Phys. and Astronomy, Brigham Young Univ., Eyring Sci. Ctr., N201, Provo, UT 84602, aaron.burton.vaughn@gmail.com), Kent L. Gee, Kevin M. Leete (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and J. M. Downing (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Crackle perception in supersonic jet noise is attributed to the presence of acoustic shocks in the waveform. This study uses an event-based beamforming method to track shock events as they propagate from the near to the far field of a high-performance military aircraft operating at afterburner. Near-field events are propagated via a nonlinear model and compared with far-field measurements. Comparisons of overall sound pressure level and spectra validate the use of the nonlinear model. The skewness of the time-derivative pressure waveform, or derivative skewness, a metric indicative of jet crackle perception, is greatly overpredicted for nonlinearly propagated waveforms. Cross-correlation coefficients of waveform segments centered about the near-field beamformed events reveal that for farther aft angles, near-field events are more related to far-field measurements. Waveform observations show that shock-like events in the near field that are more spiked in nature tend not propagate into the far field. However, near-field, large-derivative events with broader, high-pressure peaks nonlinearly steepen and form shocks in the far field that are likely responsible for crackle perception.

**11:40**

**3aPAc2. Application of event-based beamforming to the study of acoustic shocks in installed supersonic engine noise.** Jordan Grow (Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, jordan.grow@gmail.com), Aaron B. Vaughn, Kevin M. Leete (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Alan T. Wall (Air Force Res. Lab., WPAFB, OH)

Crackle is a dominant component of the noise from supersonic military-style jet engines. Because the study of crackle is tied to the characterization of acoustic shocks, examination of the locations and propagation direction of the largest-derivative waveform events in the near field can provide insights regarding source radiation and propagation characteristics. This paper describes initial results of applying an event-based beamforming algorithm to a ground-based microphone array located in the near-field of a T-7A Red Hawk aircraft with its afterburner-capable F-404 engine. The work builds on recent work by Vaughn *et al.* [AIAA Paper 2019-2664]; the algorithm identifies large-derivative events at pairs of adjacent microphones and determines the propagation direction by applying a cross correlation to short time segments around the events. Statistical analysis of these events, and associated beamforming, for the T-7A noise radiation leads to an improved understanding of the noise radiation of temporal events tied to acoustic shocks and, therefore, to crackle. [Work supported by AFRL.]

**12:00**

**3aPAc3. Examining wind noise reduction effects of windscreens and microphone elevation in outdoor acoustical measurements.** Zachary T. Jones (Phys. and Astronomy, Brigham Young Univ., Provo, UT 84602, ztj5019@psu.edu), Mylan R. Cook, Alexander M. Gunther, Taylor S. Kimball, Kent L. Gee, Mark K. Transtrum (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Shane V. Lympany, Matt Calton, and Michael M. James (Blue Ridge Res. & Consulting, Asheville, NC)

Jones *et al.* [*J. Acoust. Soc. Am.* **146,** 2912 (2019)] compared an elevated (1.5 m) acoustical measurement configuration using a standard commercial windscreen with a ground-based configuration with a custom windscreen. That study showed that the ground-based measurement method yielded superior wind noise rejection, presumably due to the larger windscreen and lower wind speeds experienced near the ground. This study examines those findings in greater depth by attempting to decouple the effects of windscreens and microphone elevation using measurements at 1.5 m and near the ground with and without windscreens. Simultaneous wind speed measurements at 1.5 m and near the ground were also made for correlation purposes. For the elevated acoustical measurements, three different commercial windscreens were used to further examine impacts of the larger windscreen in the ground-based setup. Results show that the custom windscreen has a more significant noise-reduction impact than microphone elevation, and that the ground-based setup is again preferable for obtaining broadband outdoor acoustic measurements. [Work supported by a U.S. Army SBIR.]

**12:20**

**3aPAc4. Holographic reconstructions of large-eddy simulations of a highly heated laboratory-scale jet.** Kevin M. Leete (BYU Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, KML@byu.edu), Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Junhui Liu (Naval Res. Lab., Washington, DC), and Alan T. Wall (Air Force Res. Lab., WPAFB, OH)

Near-field acoustical holography allows for the reconstruction of a sound field from a limited measurement. For aeroacoustic noise generated by military aircraft, reconstructions of the pressure field within the jet plume have provided insight into equivalent acoustic noise source distributions, though the relationship between these equivalent sources and actual flow pressures and velocities within the plume remain unknown. This study reconstructs the sound field of a highly heated laboratory-scale jet simulated by the Naval Research Lab's Jet Noise Reduction (JENRE) large eddy simulation solver. The simulated data allow for comparisons of the holographic reconstructions of the pressure and particle velocity fields to points along the nozzle lipline, to the Ffowcs Williams and Hawkings integration surface (FWHS), and to the far field. Pressure field reconstructions outside and inside of the FWHS match well, while particle velocity reconstructions follow similar trends as the velocities generated by the LES but severly underestimate the time harmonic complex amplitdues.

## Session 3aPP

# Psychological and Physiological Acoustics: Binaural Hearing (Poster Session)

Authors will be at their posters from 9:30 a.m. to 10:15 a.m.

### *Contributed Papers*

**3aPP1. Forming clusters of multiple sounds improves talker identification in an auditory scene.** William A. Yost (College of Health Solutions, ASU, PO Box 870102, ASU, Tempe, AZ 852870102, william.yost@asu.edu), M. Torben Pastore (College of Health Solutions, Arizona State Univ., Tempe, AZ), and Philip Robinson (Facebook, Redmond, WA)

In recent papers, we have shown that listeners perceive differences in the size of a small auditory scene (<4 sources) for short-duration sounds (e.g., consonant-vowel, CV, pairs) presented at about the same time, but such differences are not perceived for larger scenes. Spatial separation of a small number of sound sources affects performance much more than spatially separating a larger number of sound sources. We studied conditions when the number of sound sources is small, but the number of sounds (CVs) is larger. For example, listeners are poor at determining if a target talker is the same as a cue talker when the target is presented at one sound-source location and several distractor talkers are each presented from different sound-source locations. If all distractor talkers are "clustered (mixed)" at a single sound source spatially separated from the target sound source, then listeners are better at determining if the target talker was the same as the cue. Several additional scenarios were tested, all indicating that "clustering" sounds into a small number of sound sources improves target-talker identification compared to when sounds are presented from a larger number of sound-source locations. [Work supported by grants from NIDCD and Facebook Reality Laboratories.]

**3aPP2. Dynamic human strategies for localizing tones in rooms.** William Hartmann (Michigan State Univ., 749 Beech St., East Lansing, MI 48823, wmh@msu.edu) and Eric Macaulay (Michigan State Univ., East Lansing, MI)

Reflections and standing waves in a room cause distorted binaural information, making it difficult for listeners to localize ongoing sound sources. Possibly head motion allows a listener to sample a distorted sound field, gaining dynamic information that enables more accurate localization. The present research tries to discover listener strategies for localizing pure tones using a variable acoustics environment. Over the course of 9-s trials, a head tracker recorded head position and orientation while probe microphones in the ear canals recorded the signals and interaural differences. Therefore, the experiment provided complete information. Two model strategies are considered. In the "nulling" strategy, the listener discovers temporally and spatially discrete head pointing directions where the dominant interaural difference is zero. Candidate directions are weighted by head-pointing perseverance in the vicinity of the interaural zero. In the "inferred source" strategy the listener registers temporally and spatially continuous inferred-source candidates based on instantaneous interaural differences and head pointing directions. Each candidate is weighted by the instantaneous ratio of head-angle change to inferred-source change evaluated at the candidate location. Model strategies are evaluated by comparing weighted candidate source locations with listener response locations.

**3aPP3. An effect of eye position in cocktail party listening.** Virginia Best (Speech, Lang. and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, ginbest@bu.edu), Todd R. Jennings, and Gerald Kidd (Speech, Lang. and Hearing Sci., Boston Univ., Boston, MA)

Previous studies have noted an interaction between eye position and auditory spatial attention, including a tendency to look towards the location of an attended sound (even in the absence of useful visual information). There can also be objective improvements in the detection and discrimination of sounds when the eyes are directed to their location. In the current study, we were interested in whether there is a measurable effect of eye position in "cocktail party" listening situations. We presented five sequences of digits from five loudspeakers positioned at 0 deg, $\pm$15 deg, and $\pm$30 deg azimuth, and asked participants to repeat back the digits presented from one target loudspeaker. In different blocks of trials, the participant was instructed to visually fixate on the target loudspeaker or on a non-target loudspeaker. Head position was stabilized with a neck-rest, and eye position was tracked continuously to monitor compliance. Performance was best when eye fixation was on-target, and suffered when eye fixation was off-target, particularly for targets located in the center. This result demonstrates an influence of eye position in multitalker mixtures, even in the absence of visual information (lip-reading, etc.), and suggests that optimal performance depends on the spatial alignment of auditory and visual attention.

**3aPP4. Lateralization asymmetry in right/left ears of elderly individuals with respect to various sound pressure levels.** Kazumoto Morita (Chuo Univ., 1-13-27, Kasuga, Bunkyo-ku, Tokyo 112-8551, Japan, mtkkojiro@gmail.com), Tsukuru Osawa, and Takeshi Toi (Chuo Univ., Bunkyo-ku, Tokyo, Japan)

Authors have previously reported that elderly individuals have asymmetrical right and left ear performance when they horizontally lateralize a 1 kHz pure tone with Interaural Time Differences (ITDs). In the previous experiment, the Sound Pressure Level (SPL) was set to 60.0 dB(A), but in the present experiment, conditions were set every 1.5 dB(A) from 54.0 dB(A) to 66.0 dB(A) in order to investigate the effect of SPLs. For ITDs, 8 conditions were set to lead the right or left ear and those were 0.2, 0.4, 0.6, and 0.8 ms, respectively. The frequencies of the sounds were two types: 1 kHz and $0.5+2$ kHz. As a result, for a 1 kHz pure tone, the elderly individuals have asymmetry in right and left ear performance. As the ITD increases, it is more likely that the sound leading to the right ear is referred to as left. The lower the SPL, the more confusion exists between the right and left. Even in the elderly, confusion is rare for a sound of $0.5+2$ kHz. Young individuals rarely have confusion between the right and left ears for all conditions.

**3aPP5. Lateralization of interaurally delayed stimuli with identical amplitude spectra but different temporal envelopes.** Jörg Encke (Universität Oldenburg, Oldenburg, Germany) and Mathias Dietz (Universität Oldenburg, Küpkersweg 74, Oldenburg 26129, Germany, mathias.dietz@uni-oldenburg.de)

Interaurally delaying a 500 Hz tone by 1.5 ms is identical to advancing it 0.5 ms. Presented over headphones, humans perceive such a tone lateralized toward the side of the nominal lag. Any stimulus other than pure tones has more than one frequency component, disambiguating the true delay. When increasing the bandwidth of 1.5 ms delayed narrowband noise, the percept indeed changes from the nominal lag to the lead. However, little has been published about the stimulus- and subject specific factors that influence this changeover bandwidth. The historically first suggested mechanism requires a neural extraction of the cross-correlation function up to at least ±1.5 ms and compares its peak positions across frequency bands. Up to date this so-called "straightness" extraction is the standard model approach despite a lack of support from mammalian brainstem physiology and human brain imaging. Here, this concept is challenged by its strong suit: psychophysics. Stimuli with identical amplitude spectra but differently pronounced temporal envelopes are shown to have significantly different crossover bandwidths, despite identical straightness. The data reveal that within-channel envelope disparities—the only temporal cue at high frequencies—are also exploited at 500 Hz. The finding may help resolving model contradictions between perception and physiology.

**3aPP6. Deep learning of the binaural masking level difference.** Samuel S. Smith (School of Medicine, Univ. of Nottingham, Hearing Sci. Bldg., University Park, Nottingham NG7 2RD, United Kingdom, samuel.smith@nottingham.ac.uk) and Michael A. Akeroyd (School of Medicine, Univ. of Nottingham, Nottingham, United Kingdom)

The binaural masking level difference (BMLD) is the improvement in the detection of a signal in noise observed for different interaural configurations. Durlach [*J. Acoust. Sci. Am.* 1206–1218 (1963)] derived equations that accurately predict much of the human BMLD psychophysical data. We trained a deep neural network to predict BMLDs, as calculated with Durlach's equations, based on waveforms of a 500 Hz signal in a white noise each presented from different locations. Crucially, the network was constrained via nodes configured to embody informative representations (Iten *et al*., 2018; arXiv:1807.10300). The deep neural network accurately predicted BMLDs for stimuli within a simulated azimuth and for stimuli with interaural time differences (ITDs) outside the range of a human head. Further, even though the model was not designed to imitate neural biophysics, we discovered that the dynamics of latent nodes bore similarities with published data on neural ITD tuning and rate-level responses to BMLD stimuli. The work demonstrates how advances in deep learning can be used to consolidate theoretical and experimental approaches to binaural detection.

**3aPP7. How does temporal diffusion affect the ongoing precedence effect?** M. Torben Pastore (College of Health Solutions, Arizona State Univ., PO Box 870102, ASU, Tempe, AZ 852870102, m.torben.pastore@gmail.com) and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

In room acoustic scenarios, listeners' localization is often dominated by the sound propagating directly from the its source despite numerous reflections that present different spatial cues only milliseconds later. This is called the precedence effect (PE). Most studies have simulated the PE by presenting one sound (the lead) followed by a copy of the lead that is delayed and presented with different interaural cues (the lag). These simulations assume that reflective surfaces are flat, yet interior surfaces are often far more complex and variable, resulting in spatially and temporally diffuse reflections. The effect of the temporal aspect of this diffusion on listeners' localization of lead/lag, 200-ms duration, noise stimuli filtered to 100–900 Hz and presented over headphones is investigated. Lag stimuli are convolved with a Hanning-windowed 2-ms noise burst to simulate temporal effects of uneven reflective surfaces. Results show that listeners' localization is dominated by the interaural cues of the lead, even when gating onsets/offsets are windowed out. Modeling analyses based on those in Pastore and Braasch (2019) suggest that interaural time differences in the ongoing stimulus portion can be extracted from rising slopes of the envelopes of neural output, even when lead and lag envelopes are decorrelated.

**3aPP8. Effect of auditory spatial attention in rear side.** Ryo Teraoka (Res. Inst. of Elec. Commun., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai 860-8555, Japan, teraxtera@gmail.com), Shuichi Sakamoto, Zhenglie Cui, Yôiti Suzuki, and Satoshi Shioiri (Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Japan)

When human listeners direct the auditory selective attention to a specific direction, the listener can easily extract target sound from this direction, even under the noisy environment (commonly known as the cocktail-party effect). Although the auditory spatial attention can be directed to any directions around the listeners including rear side, it is still unclear how much the auditory spatial attention in rear side influences this phenomenon. In the present study, we investigated the effect of auditory spatial attention directing to the location behind the listener in a multi-talker environment and compared between this result and the previous result in front of the listener. For the purpose, the word intelligibility was measured for the target speech among non-target speech sounds spatially distributed around the target. The target sound was presented from the direction where the listener attended. To control the listener's attention, we manipulated the probability of target presentation directions or indicated the target direction a priori. The results revealed the improvement of word intelligibility by directing attention to a direction in the rear side, similarly to the effect of attention in the front side. This suggests that there is no directional dependency in the effect of auditory spatial attention.

**3aPP9. Reverberation detection threshold estimates in normal-hearing listeners: Effect of direct-path level.** Pavel Zahorik (Dept. of Otolaryngol. & Comm. Dis., Univ. of Louisville, Louisville, KY 40292, pavel.zahorik@louisville.edu) and Matthew T. Neal (Otolaryngol. and Commun. Disord., Univ. of Louisville, Louisville, KY)

Previous work [Zahorik, P. and Shehorn, J. *J. Acoust. Soc. Am.* **145**, 1875 (**2019**)] has used virtual auditory space techniques to measure reverberation detection thresholds in normal-hearing listeners for three sound field conditions: an office-sized room (broadband T60 = 0.5 s), a concert hall (broadband T60 = 1.5 s), and a reference condition with a single echo at 40 deg to the right of midline. Although thresholds were found to vary by as much as 18 dB across conditions, they were well predicted by a monaural temporal window model that has been used to effectively predict various temporal masking phenomena. Here, the effect of direct-path level on reverberation detection was examined for the same three listening conditions. The source signal was a 220 Hz complex tone, 250 ms in duration. In general, masking of reverberation by direct-path sound was found to increase as a function of direct path level, but only in listening conditions with shorter reverberation times. Longer reverberation times resulted in little masking at any direct path level. These results are consistent with predictions from the temporal window model, and thus further validate its use for predicting absolute sensitivity to reverberation.

**3aPP10. Impacts of component audibility and noise masking on spectral weighting of sound-localization cues.** Monica L. Folkerts (Dept. of Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. South, Nashville, TN 37232, monica.l.folkerts@vanderbilt.edu), Erin M. Picou (Dept. of Hearing and Speech Sci., Vanderbilt Univ., Nashville, TN), and G. Christopher Stecker (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE)

Spatial localization of complex sound requires appropriate integration of acoustic cues across the frequency spectrum. Spectral weighting functions (SWFs), which quantify the relative weighting of cues across frequency components, have been measured through binaural and free-field methods using observer weighting techniques adapted from Stecker and Hafter [*J. Acoust. Sci. Am.* **112**, 1046–1057 (2002)]. SWFs obtained with equal-amplitude components support a "dominance region" around 400–800 Hz [Bilsen and Raatgever, *Acoustica* **28**, 131–132 (1973); Stecker and Folkerts, *J. Acoust. Sci. Am.* **145**, 1720 (2019)]. The current study extends this work to compare the effects of component intensity on SWFs of stimuli presented

with competing noise and stimuli filtered through a high-frequency sloping hearing-loss simulation (HLS). Localization with a white-noise masker deteriorates more for low-frequency than high-frequency noise stimuli [Abel and Hay, *Scand. Audiol.* **25**, 3–12 (1996); Lorenzi *et al.*, *J. Acoust. Sci. Am.* **105**, 1810–1820 (1999)]. Imposition of spectral slopes introduced "level dominance" [Berg, *J. Acoust. Sci. Am.* **88**, 149–158 (1990); Lutfi and Jesteadt, *J. Acoust. Sci. Am.* **120**, 3853–3860 (2006)] in the form of increased weights for the loudest components. Weighting pattern changes in the presence of noise and similar patterns with the HLS due to "level dominance," if confirmed, may impact spatial hearing by shifting the dominance region. [Work supported by NIH R01-DC016643.]

WEDNESDAY MORNING, 9 DECEMBER 2020 9:30 A.M. TO 11:00 A.M.

# Session 3aSAa

## Structural Acoustics and Vibration, Physical Acoustics, Musical Acoustics, and Engineering Acoustics: Non-Contact Vibration Measurement Methods I

Tyler J. Flynn, Cochair
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Benjamin M. Shafer, Cochair
*PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406*

**Chair's Introduction—9:30**

### *Contributed Papers*

**9:35**

**3aSAa1. Numerical investigation of fully acoustic nondestructive testing method of delamination in glass fibre reinforced plastics.** Nudurupati S. Hanuman (Mech. Eng., National Inst. of Technol. Meghalaya, Bijni Complex, Shillong, Meghalaya 793003, India, nsvnhanu@gmail.com) and Tanmoy Bose (Mech. Eng., National Inst. of Technol. Meghalaya, Shillong, Meghalaya, India)

The contact based NDE methods are not suitable to inspect the structures at high elevated temperatures and large aerospace structures. Therefore, in recent days, acoustic based NDE methods are address to all these challenges. A local defect resonance (LDR) technique is one of the nonlinear techniques to get effective imaging of the defect. In this paper, the numerical work is presented for glass fibre reinforced plastic (GFRP) plate having single delamination. The abaqus software package is used to model the GFRP plate having single delamination. Explicit dynamic analysis is performed to get sound pressure levels (SPL) at defect location. Non-contact based acoustic incident wave is injected into the material through an impinging region for exciting the delamination. Impinging region is created over the one side surface of the GFRP plate, which is circular plane. Incident wave is created by using spherical co-ordinates about the center of the impinging region. Finally, optimize the spherical co-ordinates for effective excitation of delamination to get high sensitivity sound pressure levels at the delamination location. Further, the numerical work is validated with experimental work. The experiments are carried out on the GFRP plate. Corresponding sound pressure images are obtained. Finally, good judgment is achieved between the numerical and experimental work.

**9:55**

**3aSAa2. Measuring plate vibration using deflectometry: The advantages and limitations of add-on reflective material.** Gary Rhoades (Graduate Program in Acoust., Penn State Univ., Penn State University, 201 Appl. Sci. Bldg., University Park, PA 16802, glr31@psu.edu), Micah Shepherd (Acoust., Penn State Univ., State College, PA), and Jeff Harris (The Penn State Univ., State College, PA)

Deflectometry is a full field optical method which utilizes the slope fields on the surface of a planar object to track deformations. The use of a high speed camera gives the ability to measure all points on the surface instantaneously and with high spatial resolution, providing more knowledge on how structures react to transient excitations. Here, the reflected grid method will be used, as it involves geometry that allows for simple calculation of curvature distributions. This method of deflectometry relies on specular reflections to create amplifications of the measured deformations in the test plate. Therefore, this requires the plate under test to have a reflective surface. An experiment was constructed to test the vibration of a flat plate excited by an automatic force hammer. A collection of adhesive tapes, films, and spray were applied to the test object in order to increase the reflectivity of the plate. The advantages and limitations of each type of add-on reflective material will be discussed. Finally, the add-on material was used on a more complex structure to showcase the potential of deflectometry for measuring plate vibration.

10:15

**3aSAa3. Acoustical case studies of large high power hydroelectric generators: Correlation between acoustic and vibration measurements.** Francois Lafleur (Production, Hydro-QC Res. Ctr. (IREQ), 1800 boulevard Lionel-Boulet, Varennes, QC J3X 1S1, Canada, lafleur.francois@ireq.ca) and Mathieu Kirouac (Production, Hydro-QC Res. Ctr. (IREQ), Varennes, QC, Canada)

In power plants, acoustical signals are generally easier to measure than vibration signals at specific location because the security procedures are restricting the installation of accelerometers during the operation of the hydroelectric group. Specific acoustical measurements have been developed at the Hydro-Québec research center to set some maintenance actions or to gain more information about the operating conditions of the hydroelectric groups. These include acoustical airborne and structure-borne pressure measurements with conventional microphones at different locations around and inside the generator (including a microphone on the rotor) to correlate with vibration signals. Also, several vibration measurements were performed to allow ODS "Operating Deflection shape" of the generator displacement and correlation studies with the acoustic signals. Furthermore, some acoustical intensity measurements for mapping the generator floor permit the illustration of the emission pattern of the generator. All these acoustical measurements are interpreted in terms of operating conditions and specific frequencies. These acoustical and vibration measurements that were performed on several hydroelectric groups will be presented. The main goal of these measurements is to make multiphysics links between the acoustic, vibration and electromagnetic signals to perform non invasive maintenance and increase knowledge of the hydroelectric groups.

10:35

**3aSAa4. Non-contact air-coupled resonant ultrasound spectroscopy for characterization of soft structures.** Aakash Khandelwal (Dept. of Mech. Eng., Michigan State Univ., 428 S. Shaw Ln., Rm. 2555, Eng. Bldg., East Lansing, MI 48824, khande10@egr.msu.edu) and Sunil Kishore Chakrapani (Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI)

Characterization of soft, deformable materials using conventional contact-based methods can be challenging due to errors associated with the applied pressure. Therefore, techniques which use non-contact methods to characterize these soft structures are highly desirable. The present work explores the development of a non-contact air-coupled resonant ultrasound spectroscopy (AC-RUS) technique for the characterization of soft materials. A sample of polydimethylsiloxane (PDMS) was mounted as a circular membrane clamped along its periphery and was excited first in the low frequency acoustic range (i<100 Hz) to induce the lower-order resonant vibration modes. The sample was then excited at ultrasonic frequencies (>50 kHz) to excite the higher order ultrasonic modes. A preliminary assumption of linear elasticity was used to analytically model the vibration of a circular membrane, followed by validations using numerical FEM simulations. The analytical and numerical models were then modified to include the effect of viscoelasticity using Kelvin-Voigt and Maxwell models. The resonant models were further used to obtain the elastic constants and viscoelastic parameter from the experimental frequency response curve.



WEDNESDAY MORNING, 9 DECEMBER 2020 11:15 A.M. TO 12:25 P.M.

## Session 3aSAb

## Structural Acoustics and Vibration, Physical Acoustics, Musical Acoustics, and Engineering Acoustics: Non-Contact Vibration Measurement Methods II

Tyler J. Flynn, Cochair
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Benjamin M. Shafer, Cochair
*PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406*

**Chair's Introduction—11:15**

### *Contributed Papers*

11:20

**3aSAb1. Resonant ultrasound spectroscopy: Sensitivity analysis for anisotropic materials with hexagonal symmetry.** Christopher L. Sevigney (Mech. Eng., Univ. of , 3 Private Rd. 3086, Oxford, MS 38655, sevigneychris@gmail.com) and Farhad Farzbod (Mech. Eng., Univ. of MS)

Resonance ultrasound spectroscopy (RUS) is an experimental method for measuring the elastic properties of a material. A sample is excited to vibrate, and its resonant frequencies are measured. From the resonant frequencies and mode shapes, a complete set of elastic constants can be extracted. This result is not always reliable, however. In some cases, the resonant frequencies are insensitive to changes in certain elastic constants or their linear combinations. Previous work has been done to characterize these sensitivity issues in materials with isotropic and cubic symmetry. This work examines the sensitivity of elastic constant measurements by the RUS method for materials with hexagonal symmetry, such as titanium-diboride.

We investigate the reliability of RUS data and explore supplemental measurements to obtain an accurate and complete set of elastic constants.

**11:40**

**3aSAb2. Single-sensor remote damage localization in vibrating structures using compressive sensing.** Tyler J. Flynn (JHU/APL, 11100 Johns Hopkins Rd., Laurel, MD 20723-6099, tyler.flynn@jhuapl.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Many engineering structures are subject to broadband excitation while in operation, and it is common for such structures to radiate measurable sound as a result. When a structure is damaged the character of its frequency response changes and these changes can be exploited for health monitoring. In particular, shifts in resonant frequencies are appealing indicators of damage because they are relatively easy to obtain even with a single, non-contacting sensor. Previous methods have been developed to localize structural damage by minimizing an objective function of the differences between measured modal frequencies and those computed from an analytical model (e.g., a finite element model). However, these approaches suffer significantly when measured resonances are not known exactly, or when only few modes are obtainable. This talk details a compressive sensing (CS) approach to the damage localization problem that yields improved localization accuracy when only a small set of modal frequencies is available, even with imprecision. Simulated results are presented for both 1D (beam) and 2D (plate) systems using analytical finite element models, comparing the conventional and CS approaches. Effects of sensor noise and model inaccuracy are addressed. [Sponsored by NAVSEA through the NEEC and by the US DoD through an NDSEG Fellowship.]

**12:00**

**3aSAb3. Numerical modeling of ultrasonic leaky Lamb wave propagation in screen pipes.** Xinyue Gong (Dept. of Phys. and Astronomy, The Univ. of MS, 145 Hill Dr., University, MS 38677, xgong@go.olemiss.edu)

Screen pipes completion is often used in unconsolidated sandstone reservoirs to prevent formation sand from entering wellbore. However, the screen pipes can be blocked by the formation sand, leading to reduced output of oil. Here a technology based on attenuation of ultrasonic leaky Lamb waves is used to evaluate the sand accumulation outside the screen pipes. A three-dimensional parallel finite-difference time-domain numerical method is developed to investigate the attenuation in the case of a screen pipe surrounded by water or sediment. The attenuation pattern determines the presence of screen holes and the acoustic impedance of the medium outside the screen pipes. The study demonstrates the feasibility of the leaky Lamb wave technology to evaluate sand accumulation outside screen pipes.

WEDNESDAY MORNING, 9 DECEMBER 2020 — 9:30 A.M. TO 11:00 A.M.

## Session 3aSCa

## Speech Communication, Architectural Acoustics, Psychological and Physiological Acoustics, and Noise: Listening in Challenging Circumstances I

Kristin J. Van Engen, Cochair
*Psychological and Brain Sciences, Washington University in St. Louis, One Brookings Pl, Campus Box 1125, St. Louis, MO 63130*

Melissa M. Baese-Berk, Cochair
*University of Oregon, 1290 University of Oregon, Eugene, OR 97403*

**Chair's Introduction—9:30**

### *Invited Papers*

**9:35**

**3aSCa1. Listening challenges for service members and veterans with traumatic brain injury: The role of sentence context for speech perception in noise.** Stefanie E. Kuchinsky (Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., 4954 North Palmer Rd., Bldg. 19, Fl 5, Bethesda, MD 20889, skuchins@umd.edu), Megan M. Eitel (Audiol. and Speech Pathol., Walter Reed National Military Medical Ctr., Bethesda, MD), Julie I. Cohen (Hearing and Speech Sci., Univ. of Maryland, College Park, MD), Rael T. Lange, Louis M. French, Tracey A. Brickell, Sara M. Lippa (Defense and Veterans Brain Injury Ctr., Bethesda, MD), and Douglas S. Brungart (Audiol. and Speech Pathol., Walter Reed National Military Medical Ctr., Bethesda, MD)

Service Members and Veterans (SMVs) with a traumatic brain injury (TBI) often report difficulties understanding speech in noise. This study assessed the impact of TBI history on speech-in-noise recognition and benefit from sentence context. Participants were SMVs (ages 20–62) who had a history of at least an uncomplicated mild TBI ($n = 122$) or who had no TBI history ($n = 58$). For each listener, an interleaved, adaptive procedure was used to obtain an overall Speech Reception Threshold (SRT) and a difference in accuracy for high and low context sentences at that threshold. Penalized LASSO regressions revealed larger context benefits for individuals with

TBI, poorer high frequency thresholds, poorer executive function, and self- reported tinnitus. Age, Post-Traumatic Stress Disorder, and neurobehavioral symptom severity did not predict context benefit. These results suggest TBI is associated with greater reliance on context in noise, even when accounting for variation in hearing acuity and cognitive function. [This project was funded by the Defense and Veterans Brain Injury Center (DVBIC) as part of a Congressionally-Mandated Longitudinal TBI Study. 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.]

**9:55**

**3aSCa2. Examining bilingual speech perception and the role of background noise.** Giovanna Morini (Commun. Sci. and Disord., Univ. of Delaware, 100 Discovery Blvd, Newark, DE 19713, gmorini@udel.edu)

Over the years there has been much debate regarding how bilingualism influences performance on linguistic tasks. One specific question is whether growing up with one versus two languages is related to variations in the ability to process speech in adverse listening conditions. This talk will include findings from studies with both adults and young children that compared monolinguals' and bilinguals' ability to recognize and learn words in the presence of white noise. The adult data suggest that bilinguals are less accurate than monolinguals at identifying familiar words in noise. However, the bilingual "disadvantage" identified during word recognition is not present when listeners were asked to acquire novel word-object relations that were trained either in noise or in quiet. Similar group differences were identified with 30-month-olds. Bilingual children performed significantly worse than monolinguals, particularly when asked to recognize words that were accompanied by noise. This work suggests that linguistic experience and the demands associated with the type of task both play a role in the ability for listeners to process speech in noise. Furthermore, it suggests that bilingualism not only plays an important role in speech processing in noise, but that this effect is preset from a very early age.

**10:15**

**3aSCa3. Measuring the subjective cost of listening effort using a discounting task.** Drew J. McLaughlin (Psychol. & Brain Sci., Washington Univ. in St. Louis, 1 Brookings Dr., St. Louis, MO 63130, drewjmclaughlin@wustl.edu), Todd Braver (Psychol. & Brain Sci., Washington Univ. in St. Louis, St. Louis, MO), and Jonathan Peelle (Otolaryngol., Washington Univ. in St. Louis, St. Louis, MO)

Objective measures of listening effort have been gaining prominence, as they provide metrics to quantify the difficulty of understanding speech. A key challenge has been to develop paradigms that enable the complementary measurement of subjective listening effort in a quantitatively precise manner. In the present study, we used a novel decision-making paradigm to examine age-related and individual differences in subjective effort during listening for spoken sentences in speech-shaped noise. On each trial subjects were offered the choice between completing an easier trial (presented at a +20 dB signal-to-noise ratio; SNR) for a smaller monetary reward, or a harder trial (presented at either +4, 0, −4, −8, or −12 dB SNR) for a greater monetary reward. By varying the amount of the reward offered for the easier option, the subjective value of performing effortful listening trials at each SNR could be assessed. Older adults discounted the value of effortful listening to a greater degree than young adults, opting to accept less money in order to avoid more difficult SNRs. Additionally, older adults with poorer hearing and smaller working memory capacities were more likely to choose easier trials; however, in younger adults, no relationship with hearing or working memory was found.



**10:35**

**3aSCa4. Predicting children's word recognition accuracy from accent distance metrics.** Tessa Bent (Speech, Lang. and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, tbent@indiana.edu) and Rachael F. Holt (Speech and Hearing Sci., Ohio State Univ., Columbus, OH)

Compared to adults, children show lower word identification accuracy under both environmental- and source-related listening challenges (e.g., speech in noise or unfamiliar accents). Early school-aged children have difficulty understanding both unfamiliar native and nonnative accents. Although adults typically have difficulty only when unfamiliar accents are mixed with noise, children show word identification decrements for some unfamiliar accents under quiet conditions. Furthermore, children's word recognition accuracy varies widely across accents in both quiet and noise-added conditions. The developmental differences appear to stem partially from children's less robust use of semantic/syntactic contextual cues; less well understood is the role of specific accent characteristics leading to reduced word identification. We are currently assessing the relation between a number of accent distance metrics and children's word recognition accuracy. These metrics, which quantify the distance from the home dialect, include segmental (i.e., Levenshtein distances), holistic signal (i.e., dynamic time warping), holistic perceptual (i.e., listener rankings from the native dialect standard), and suprasegmental (e.g., articulation rate, speech rhythmic metrics) measures. This work may provide insight into what aspects of the signal cause children more difficulty than adults when mapping unfamiliar pronunciations onto their lexical representations. Work supported by the National Science Foundation, Grant No. 1941691.]

## Session 3aSCb

# Speech Communication, Architectural Acoustics, Psychological and Physiological Acoustics, and Noise: Listening in Challenging Circumstances II

Kristin J. Van Engen, Cochair
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Melissa M. Baese-Berk, Cochair
*University of Oregon, 1290 University of Oregon, Eugene, OR 97403*

**Chair's Introduction—11:15**

## *Invited Papers*

**11:20**

**3aSCb1. The mental cost of repairing errors in speech perception.** Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Shevlin Hall Rm. 115, Minneapolis, MN 55455, mwinn@umn.edu) and Katherine Teece (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

When evaluating a listener's ability to hear speech in a challenging condition, an experimental challenge is determining whether an utterance was heard correctly or whether it was heard incorrectly and then mentally repaired. Mentally transforming incorrect perceptions into "correct" responses is not visible in intelligibility scores, but is likely a major part of what makes listening effortful for people with hearing loss, and therefore there is a need to design tests that track the occurrence and mental cost of making corrections during speech perception. A series of studies is presented which gathers evidence of the mental cost of repairing errors using pupillometry. Stimuli were explicitly designed to induce different types of cognitive repair so that the ensuing mental cost/listening effort can be measured as it unfolds over time, even without major changes in intelligibility. Results show that different errors result in significantly different amounts of effort. Major findings include quick bursts of effort for phonetic repairs, larger prolonged bursts of effort for entire-word repairs constrained by context, and a substantial increase in effort for errors that result in semantic incongruity across an utterance. Results are discussed with regard to implications for clinical interpretation as well as study design.

**11:40**

**3aSCb2. Listening effort assessed using engaging, naturalistic materials.** Ingrid Johnsrude (Psychology/Comm Sci. Disord., Univ. of Western Ontario, 4124 Western Interdisciplinary Res. Bldg., London, ON N6A 5B7, Canada, ijohnsru@uwo.ca), Matthew T. Bain (Neurosci., Univ. of Western , London, ON, Canada), Aysha Motala (Psychology/Comm Sci Disord, Univ. of Western Ontario, London, ON, Canada), and Björn Herrmann (Rotman Res. Inst., Toronto, ON, Canada)

Hearing loss in older people is typically diagnosed long after they begin to find speech comprehension effortful in the presence of background sound. Progress in measuring listening effort has been slow because the concept is ill defined, the materials typically used to measure it (simple sentences) may not motivate effortful listening the way that richer narratives do, and the cognitive abilities and brain networks that are most related to listening effort have not been systematically identified. We are examining the utility of engaging, naturalistic stories, compared lo isolated sentences, to measure listening effort with novel behavioural and functional magnetic resonance imaging (fMRI) methods that provide a window on the cognitive processes recruited to compensate for masked speech. We exploit the fact that naturalistic and engaging stories are known to activate much of the brain, in specific patterns that are timelocked to a story. The goal of the research is to develop a clearer understanding of the nature of listening effort, and begin to identify relevant markers in cognition and brain network activity, so that we can make more rapid progress on development of sensitive tests that will enable more timely, efficient, and beneficial fitting of hearing aids.

## Session 3aSPa

# Signal Processing in Acoustics: General Topics in Signal Processing in Acoustics III

Efren Fernandez-Grande, Cochair
*Copenhagen, Denmark*

Eric A. Dieckman, Cochair
*Mechanical Engineering, University of New Haven, 300 Boston Post Road, West Haven, CT 06516*

**Chair's Introduction—9:30**

## *Contributed Papers*

**9:35**

**3aSPa1. Passive velocity estimation of remotely-controlled vehicles via Doppler tracking.** Eden Oelze, Jonah Singer (Mahomet Seymour High School, Mahomet, IL), and Gizem Tabak (Univ. of Illinois at Urbana-Champaign, 1308 W Main St., 119 Coordinated Sci. Lab., Urbana, IL 61801, tabak2@illinois.edu)

The performance of remotely-controlled (RC) vehicles in recreational activities, such as RC cars, boats, planes and drones, has increased dramatically with the increased energy density of lithium polymer and Nickel metal hydride battery technologies. As a result, RC cars capable of land speeds in excess of 100 mph are available in hobbyist-class vehicles. This work presents an experimental and analytical setup for measuring the land-, water-, and air-speed of such RC vehicles using passive signal processing of acoustic recordings of the vehicles in operation. The high-efficiency DC brushless motors used in these platforms emit strong harmonic structure that can be efficiently measured with Doppler-tracking. The harmonic structure of the recorded acoustic signals allow passive velocity estimation using quasi-periodic signal detection and period estimation techniques based on pitch detection methods in the time and frequency domain. Preliminary results yielded successful velocity recovery based on Doppler tracking using a pilot signal, and demonstrated the correlation between the speed profile of the vehicle and acoustic harmonics. Future work will include an analytical model and set of experiments for passive velocity measurement suitable for high school and undergraduate physics laboratory exercises.

**9:55**

**3aSPa2. Investigating the effect of off-axis bearing estimation on the range-Doppler characteristics of transmit waveforms.** Matthew Tidwell (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, matthew.d.tidwell@navy.mil) and David A. Hague (Naval Undersea Warfare Ctr., Newport, RI)

The spectral filtering an active sonar waveform undergoes due to the frequency dependent beampatterns of a projector encodes information about the target's bearing relative to the main response axis. For a given transducer and transmit spectrum there exists an optimal angle of operation at which the Fisher Information (FI) of the target bearing estimate is maximized. For relatively low fractional bandwidth Linear Frequency Modulated (LFM) waveforms, the angle of maximum FI is solely determined by its center frequency [Tidwell and Buck, in SSPD (2019)]. This facilitates steering the region of maximum bearing estimation precision by changing the center frequency of the LFM waveform without steering the transducer's main response axis. This research expands on these previous efforts and investigates both the transmit waveform impact on bearing estimation performance and the spectral filtering impact on the waveform's range-Doppler ambiguity function. This analysis is performed using the Multi-Tone Sinusoidal Frequency Modulated (MTSFM) waveform model, which possesses a discrete set of parameters that are adjusted to synthesize a broad class of waveform types [Hague, in IEEE AES (2020)]. Low fractional bandwidth MTSFM waveforms have a negligible impact on bearing estimate precision. However, the transducer's spectral filtering has a profound impact on the waveform's range-Doppler characteristics.

**10:15**

**3aSPa3. A Bayesian dynamic time warping approach for nonuniform Doppler paths in acoustic multipath channel.** Sijung Yang (Elec. and Comput. Eng., Univ. of Illinois at Urbana-champaign, 1308 West Main St., Coordinated Sci. Lab., Urbana, IL 61801, Syang103@illinois.edu), Jae Won Choi, and Andrew C. Singer (Elec. and Comput. Eng., Univ. of Illinois at Urbana-champaign, Urbana, IL)

The estimation of Doppler effects is indispensable to many acoustic signal processing applications including reliable underwater acoustic communication. Computationally expensive ambiguity-function based methods have good performance in estimating slowly varying uniform Doppler shifts. However, for highly reverberant environments such as in shallow water acoustics, there has not been a robust approach to estimate non-uniformly time varying Doppler along each of the different propagation paths. In this paper, we propose a multidimensional dynamic time warping (DTW) algorithm to precisely estimate non-uniform time varying Doppler in acoustic multipath channels. The proposed algorithm exploits a Bayesian approach to map multidimensional temporal scaling, while guaranteeing polynomial time complexity in the length of the signal. Both simulation and experimental results for signals received from moving sources in confined reverberant spaces will be provided to show the capability of the proposed approach.

**10:35**

**3aSPa4. Doppler tracking and compensation for underwater acoustic channels using shift-orthogonal pilot sequences.** Ali Bassam (Elec. and Comput. Eng., Dalhousie Univ., 1360 Barrington St., Halifax, NS B3H 4R2, Canada, al903300@dal.ca) and Mae Seto (Elec. and Comput. Eng., Dalhousie Univ., Halifax, NS, Canada)

A novel time-domain Doppler tracker and compensator, using shift-orthogonal OFDM pilot sequences, was developed, and assessed in simulations and early in-water trials. The objective is to address large Mach numbers like those experienced in communicating autonomous underwater vehicles (AUV). To start, the Doppler estimator extracts pilots in the received signal to produce Mach number estimates for every received signal sample. Given the large number of estimator samples, a Doppler tracker is proposed to reduce the number and simultaneously track Mach number

3a WED. AM

variations. This novel tracker reduces the computational load on the compensator and makes it possible to implement on AUVs. Then, the Doppler compensator's first stage resamples the received signal using the Mach number estimates from the tracker's output. Since the estimates vary with time, the resampling performed is time-varying. Then, the final stage estimates the residual Doppler shift from the estimator and eliminates it with a phase rotation. The proposed tracker and compensator receiver outperform most existing compensators as measured through the mean-squared error, is more computationally efficient and addresses higher Mach numbers. Each element of the proposed receiver is tested in simulations and the results are shown to agree with theory. Comprehensive at-sea trials are next.

WEDNESDAY MORNING, 9 DECEMBER 2020 11:15 A.M. TO 12:05 P.M.

## Session 3aSPb

# Signal Processing in Acoustics, Animal Bioacoustics, Engineering Acoustics, Underwater Acoustics, and Acoustical Oceanography: Acoustic Localization V

Zoi-Heleni Michalopoulou, Cochair
*Department of Mathematical Sciences, New Jersey Institute of Technology, 323 M. L. King Boulevard, Newark, NJ 07102*

Paul J. Gendron, Cochair
*ECE, Univ. of Massachusetts Dartmouth, 285 Old Westport Rd, Dartmouth, MA 02747*

Kainam T. Wong, Cochair
*School of General Engineering, Beihang University, Beihang University, New Main Building D-1107, 37 Xueyuan Road, Beijing, 100083, China*

**Chair's Introduction—11:15**

### *Contributed Papers*

**11:20**

**3aSPb1. Inverse laplace transform and image-based room impulse response for information on source location.** ambika bhatta (Elec. and Comput. Eng., Physical and Life Sci. Solutions LLC, UMass Lowell, Lowell, MA 01854, ambikabhatta@gmail.com), Pratik Gandhi (Elec. and Comput. Eng., UMass Lowell, Lowell, MA), and Max Denis (Univ. of the District of Columbia, Washington, DC)

The source localization in indoor and outdoor using sensors based empirical model has been traditionally useful. The presented work is investigating the possibility of extending an exact "image-based" analytical room impulse response (RIR) to classify the source region. The approach is governed by its validity for a spherical acoustic simple source. One technique for a given spatial grid the time responses being mapped to an intensity profile for grayscale images will be processed by a tailored neural network, and other, the conventional processing of raw impulse responses to obtain a likelihood function. The study for both the outcomes will be of particular interest to determine location-driven features.

**11:40**

**3aSPb2. Using frequency-differencing to robustly localize distant sources in the deep ocean.** David J. Geroski (Appl. Phys., Univ. of Michigan, 2313 Packard St., Apt A103, Ann Arbor, MI 48104, geroskdj@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Frequency-Differencing (FD) methods have proven successful for localizing distant sources both in the deep ocean [Geroski and Dowling, *J. Acoust. Sci. Am.* **146**, 4727–4739 (2019)] and in the shallow ocean [Worthmann *et al.*, *J. Acoust. Sci. Am.*, **138**, 3549–3562 (2015)]. These source localizing algorithms are based upon matching the phase of a product of measured complex field values, termed autoproducts, to a replica field that is calculated based on the user's knowledge of the environment, similar to Matched Field Processing (MFP). Unlike MFP, FD methods have proven to be robust to the problem of environmental mismatch in ocean environments. Because of the focus on demonstrating robustness to environmental mismatch, these efforts have provided little guidance for a user to apply such methods in the practice. This presentation explores the process of how to choose difference frequencies and difference frequency bandwidths to localize sources at long ranges in the deep ocean, as well as how to use snapshot averaging to improve FD source localization results. These results and limitations are demonstrated using measurements taken in the PhilSea10 experiment from moored sources as far as 450 km away from the vertical receiving array, as well as simulations mimicking acoustic propagation in this environment. [Sponsored by ONR.]

# Session 3aUW

## Underwater Acoustics: Sources of Underwater Sound

Derek Olson, Chair
*Oceanography, Naval Postgraduate School, Spanagel Hall, 833 Dyer Road, Monterey, CA 93943*

**Chair's Introduction—9:30**

### *Contributed Papers*

**9:35**

**3aUW1. Long term acoustic time series of the Lofoten–Vesterålen ocean observatory.** Geir Pedersen (Marine Ecosystem Acoust., Inst. of Marine Res., Fantoftvegen 38, Bergen 5892, Norway, gepe@norceresearch.no), Guosong Zhang (Marine Ecosystem Acoust., Inst. of Marine Res., Bergen, Norway), Sofia Aniceto (UiT The Arctic Univ. of Norway, Tromsø, Norway), and Espen Johnsen (Marine Ecosystem Acoust., Inst. of Marine Res., Bergen, Norway)

Lofoten-Vesterålen (LoVe) is a productive coastal shelf-slope area and an environment sensitive to external stressors. This is a region with complex dynamics where the Norwegian Coastal Current and the Norwegian Atlantic Slope Current meet and mix, and commercially and ecologically important species of fish go through early vulnerable life stages drifting through during planktonic life stages. The LoVe Ocean Observatory is a cabled multi-purpose observation network in development, where the first node was installed and has been in operation since 2013. The main objective of the observatory is to significantly increase knowledge of the physical, chemical, and biological environment of the LoVe shelf-slope-system. The observatory nodes are equipped with a range of sensors including active and passive acoustic sensors. Echosounders monitor vertical distribution and density of marine organisms and flux of biomass across the observatory transect. Hydrophones provide continuous monitoring of anthropogenic noise and vocalizing marine mammals and fish. In this work we examine the active and passive acoustic time series from the first years of operation of the observatory, with emphasis on periods of co-occurrence of lower trophic level organisms and top predators.

**9:55**

**3aUW2. Underwater sound source characteristics from down-the-hole pile driving.** Shane Guan (Office of Protected Resources, National Marine Fisheries Service, 1315 East-West Hwy., SSMC-3, Ste. 13700, Silver Spring, MD 20902, shane.guan@noaa.gov) and James Reyff (Illingworth & Rodkin, Inc., Cotati, CA)

Unlike conventional impact pile driving technique in which a pile is hammered into the sediment a top hammer, down-the-hole (DTH) piling technique uses a combination of percussion and drilling mechanism with the hammer acting directly on the sediment to advance the pile. Small cuttings and debris broken by the hammer are flushed out by the air exhaust through the drill pipes. DTH piling is considered one of the fastest ways to drill through hard rock and install piles, and is increasingly used by industries. However, underwater noise generated from DTH piling is poorly studied. Here we present two DTH pile driving studies: pile driving of 18- and 42-inch steel pipe piles on Biorka Island and at Skagway, Alaska, respectively. The results show that single strike sound exposure levels are 146 and 164 dB *re* 1 $\mu$Pa$^2$ s at 10 m for the 18- and 42-in. piles, respectively, the root-mean-square sound pressure levels at this distance are 162 and 178 dB *re* 1 $\mu$Pa for the 18- and 42-in. piles, respective piles. Our results are compared to those from other studies. The potential effects on marine fauna resulting from this complex sound field needs further investigation.

**10:15**

**3aUW3. Model for explosive shock waves in fluids with relaxation.** William A. Willis (Appl. Res. Labs., The Univ. of Texas at Austin, TX 78713-8029, william.willis@utexas.edu), John M. Cormack Dept. of Medicine, (Ctr. for Ultrasound Molecular Imaging and Therapeutics, Univ. of Pittsburgh, Pittsburgh, PA), Thomas G. Muir, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Nonlinear propagation of sound from underwater explosions has been treated empirically and analytically since the work of Cole [*Underwater Explosions* (1948)]. A semi-analytical solution for propagation of a weak shock followed initially by an exponential tail was obtained by Rogers [*J. Acoust. Sci. Am.* (1977)] under the assumption of no relaxation. Here, an efficient algorithm is presented for calculating the propagation of a weak shock followed by a tail with arbitrary waveform in a fluid with multiple relaxation mechanisms. The model is based on an evolution equation in intrinsic coordinates for nonlinear propagation in a relaxing fluid [Hammerton and Crighton, *JFM* (1993)]. Intrinsic coordinates permit the waveform to become multivalued, which avoids having to discretize thin shocks. At distances where output is desired, a shock is inserted according to Landau's equal-area rule, rendering the waveform single-valued. Waveforms calculated in this way agree with numerical solutions of a Burgers equation augmented to include relaxation, with the proposed method requiring less computation time. When an exponential tail following the shock decays faster than the shortest relaxation time, explicit solutions are obtained for the amplitude and arrival time of the shock. [W.A.W. is supported by the ARL:UT Chester M. McKinney Graduate Fellowship in Acoustics.]

**10:35**

**3aUW4. Bubble curtain effectiveness during impact pile driving for monopile installation at the Coastal Virginia Offshore Wind project.** Jennifer L. Amaral (Marine Acoust., Inc., 2 Corporate Pl., Ste. 105, Middletown, RI 02842, jennifer.amaral@marineacoustics.com), Adam S. Frankel (Marine Acoust., Inc., Middletown, RI), James H. Miller, Gopu Potty (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Ying T. Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, RI), Arthur E. Newhall (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), and Tim Mason (Subacoustech Environ., Ltd., Southampton, United Kingdom)

The Coastal Virginia Offshore Wind (CVOW) pilot project consists of two 6-megawatt wind turbines located 27 miles off the coast of Virginia Beach, VA. Monopile foundations with a diameter of 7.8 m at the seafloor were installed via impact pile driving on two separate days during May 2020. A double bubble curtain was used during the installation of one of the monopiles and no sound mitigation system was used during the installation of the second. The resulting acoustic field was measured during the impact

pile driving using a suite of stationary and towed sensors to characterize the effectiveness of the bubble curtain in attenuating the sound levels at various ranges and azimuths. Metrics including the peak sound pressure level, sound exposure level, and kurtosis of each pile strike were determined and analyzed as a function of distance from the foundation. The frequency and azimuthal dependence of the bubble curtain effectiveness was also investigated. [Work supported by Bureau of Ocean Energy Management (BOEM.)]

WEDNESDAY AFTERNOON, 9 DECEMBER 2020 1:05 P.M. TO 2:35 P.M.

# Session 3pAAa

## Architectural Acoustics, Noise, Education in Acoustics, and Speech Communication: New Developments in Classroom Acoustics I

Laura C. Brill, Cochair
*Threshold Acoustics, 141 W Jackson Blvd, Suite 2080, Chicago, IL 60604*

David Lubman, Cochair
*DL Acoustics, 14301 Middletown Ln., Westminster, CA 92683*

David S. Woolworth, Cochair
*Roland, Woolworth & Associates, 356 County Road 102, Oxford, MS 38655-8604*

**Chair's Introduction—1:05**

### *Invited Paper*

**1:10**

**3pAAa1. Variable refrigerant flow HVAC units in classrooms—Calculations versus measurements.** David Manley (DLR Group, 6457 Frances St., Omaha, NE 68106, dmanley@dlrgroup.com) and Logan Pippitt (DLR Group, New York, NY)

An elementary school renovation and addition design seeking LEED certification included use of a variable refrigerant flow system for classroom heating and cooling. Previous challenges have been identified with use of these types of units, most notably that sound data provided by equipment manufacturers is non-standardized to AHRI or other United States or internationally recognized HVAC sound data collection procedures. Methods for estimating corrections to the non-standardized noise data developed by others were deployed to estimate room noise levels in design phases, but compliance with LEED noise level requirements of 40 dBA was not able to be satisfactorily determined. Post-construction field measurements were conducted, and results will be presented and compared to design calculations.

### *Contributed Paper*

**1:30**

**3pAAa2. Achieving CHPS for classroom acoustics: Project case study.** William Rosentel (Acoust., TEECOM, 1333 Broadway, Oakland, CA 94612, william.rosentel@teecom.com) and Peter Holst (Acoust., TEECOM, Oakland, CA)

This presentation will provide insights from a recently completed, budget-driven new construction high school project in California, which pursued the CHPS (Collaboration for High Performance Schools) Designed program. The nation-wide CHPS criteria include acoustical performance pre-requisites for project recognition under the indoor environmental quality metric. The acoustical criteria address sound isolation, room acoustics, HVAC background noise levels, and environmental noise intrusion. The pre-requisite acoustical criteria applied to this project (EQ 14.0—2014 CA-CHPS Criteria v1.02) will be reviewed along with discussion of the optional EQ 14.1 Enhanced Acoustical Performance. The CHPS criteria reference ANSI 12.60 and describe testing methods regarding compliance evaluation. However, interpretation of the criteria is still required. Improvement to the criteria language and testing requirements are of pragmatic interest to achieving the CHPS design intent. The presentation will overview the required documentation provided during the Design Phase and a summary of acoustical recommendations to achieve the criteria, which have cost implications and required coordination across design team members. Review of the final design, post-construction testing and findings will be provided, highlighting the components of the acoustical CHPS criteria that were readily achieved, and those components where implementation of the design must be carefully overseen during construction.

## *Invited Papers*

**1:50**

**3pAAa3. Implications of modern classroom architectural design and pedagogy on meeting ANSI S12.60 national classroom acoustics standard requirements.** David s. Woolworth (Roland, Woolworth & Assoc., 356 County Rd. 102, Oxford, MS 38655-8604, dwoolworth@rwaconsultants.net)

$21^{st}$ century architectural design includes familiar acoustical challenges such as daylighting, but also includes flexible spaces that recall the modern multi-environment office design and their architectural minimalism. This investigation examines the core elements of the modern classroom design and the resulting acoustical design challenges to meet the national classroom standard. The data, decision processes, and solutions presented are taken from a school currently under construction.

**2:10**

**3pAAa4. STEM in action: One progressive school's project-based approach to improving their acoustical environment.** Sean Browne (Innovation, Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17603, sdbrowne@armstrongceilings.com) and Spencer Browne (The Stone Independent School, Lancaster, PA)

They go by many names: Progressive Schools, $21^{st}$ Century Schools, Project-Based Schools. These schools do not conform to the test-driven curricula often mandated by local school boards but have student-centered curricula with an emphasis on developing portfolios and exhibiting tangible products as evidence of learning. These schools are rapidly increasing in numbers and leading an evolution in education. The schools that practice project-based learning have more laboratories and maker spaces. They will be more hands-on in nature and very collaborative. They have dynamic needs from the physical space, having to accommodate individual, small-group, and large-group activities. However, they will not be immune from the detriments of excessive background noise and reverberation. The nature of these types of schools will present challenges to the acoustical designs that are needed to ensure that "learning is easier, deeper, more sustained, and less fatiguing." A case study will be presented of such a school and how their need for an acoustical remediation became their class project.

## Session 3pAAb

# Architectural Acoustics, Noise, Signal Processing in Acoustics, and Engineering Acoustics: Session in Memory of Jiri Tichy III

Victor W. Sparrow, Cochair
*Penn State, 201 Applied Science Bldg., University Park, PA 16802*

Gary W. Elko, Cochair
*mh acoustics, 25A Summit Ave., Summit, NJ 07901*

**Chair's Introduction—1:05**

### *Invited Papers*

**1:10**

**3pAAb1. Dr. Jiri Tichy: A man missed not only for his technical contributions to science but for the personal support and encouragement he gave to his graduate students.** John Parkins (Red Tail Hawk Corp., 1111 Locust St., Unit 8i, Philadelphia, PA 19107, jparkins@rthcorp.com)

Dr. Tichy was my co-advisor during my doctoral work at Penn State University. This work involved architectural acoustics, three-axis energy density measurements and feedforward active noise control in rooms. I was also a student of Dr. Tichy's, as I took his course on acoustic intensity. From the time I met Dr. Tichy, during my application interview for the graduate program, to the time I left the program, after a postdoctoral position in thermoacoustics, Dr. Tichy was not only supportive from a technical perspective but also from a personal one. In this presentation, I'd like to touch on Dr. Tichy's technical work, his impact on me as a researcher and also how he was an inspiring, warm and supportive mentor.

**1:30**

**3pAAb2. Final Master's project advised by Jiri Tichy on architectural acoustics.** Alexandra Loubeau (NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, a.loubeau@nasa.gov)

Jiri Tichy's interest and expertise in architectural acoustics research led to the formulation of a Master's student project on the irregularity of low-frequency response in small rooms. The numerical study conducted under the guidance of Dr. Tichy by the present author addressed how to decrease the low-frequency sound field variability using a loudspeaker source with constant sound power radiation. A summary of the research results will be presented in terms of standard deviation, as a function of source position, receiver position, and room absorption. In addition, the presentation will include how this research experience instilled in the author a desire to continue pursuing acoustics research.

**1:50**

**3pAAb3. From statistical room acoustics and acoustic intensity to spatial audio.** Gary W. Elko (mh Acoust., 25A Summit Ave., Summit, NJ 07901, gwe@mhacoustics.com)

Jiri Tichy was my graduate student advisor at Penn State. Jiri introduced me to statistical room acoustics and sound power measurements using a single pressure microphone in a reverberation room. From there we moved on to the measurement of sound power using acoustic intensity. We developed spatial microphone array probes with 2–4 microphones that measured the complex vector acoustic intensity in one to three dimensions. After graduation, the fundamentals developed analyzing the estimation of the complex acoustic intensity were applied to the development of a spherical microphone array that realized any first-order superdirectional directivity pattern pointing in any direction. It later became clear that higher-order arrays were needed to achieve higher directional gain. This led to the development of the Eigenmike® spherical microphone array with 32 pressure microphones and capable of 4th order spherical harmonic decomposition of the sound field. This array is finding application into AR/VR and statistical room acoustic analysis … bringing the technology full circle. I am incredibly lucky and grateful to have had Jiri as my mentor. He showed me how to look at acoustics problems and set the foundations of the many acoustic signal processing areas I have worked on in my career.

**2:10–2:30**

## Session 3pAAc

# Architectural Acoustics, Noise, Education in Acoustics, and Speech Communication: New Developments in Classroom Acoustics II

Laura C. Brill, Cochair
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David Lubman, Cochair
*DL Acoustics, 14301 Middletown Ln., Westminster, CA 92683*

David S. Woolworth, Cochair
*Roland, Woolworth & Associates, 356 County Road 102, Oxford, MS 38655-8604*

**Chair's Introduction—2:50**

## *Invited Papers*

**2:55**

**3pAAc1. Speech intelligibility in the primary classroom: What if the teacher has voice problems.** Pasquale Bottalico (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 South Sixth St., Champaign, IL 61820, pb81@illinois.edu) and Silvia Murgia (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

School children need clear auditory signals and low background noise to learn. When classroom acoustics are poor, teachers often compensate by raising their voices, usually with limited effect against background noise, and, long-term, this makes vocal overuse the primary cause (60%) of the high prevalence of voice problems in teachers. Speech intelligibility tests were performed in primary schools with normal hearing students using words produced by an actor with normal voice quality and simulating a dysphonic voice. The speech was played by a Head and Torso Simulator. Artificial classroom noise and classrooms with different reverberation times were used to obtain a range of Speech Transmission Index from 0.2 to 0.7 (from bad to good). Results showed a statistically significant decrease in intelligibility when the speaker was dysphonic with a maximum of 15% intelligibility loss. This study extends an important pairing of problems related to student learning: classroom acoustics and teachers with voice disorders. It provides important insights into the enormous variability in speech intelligibility in classrooms by characterizing students' intelligibility when students receive degraded auditory input. The degraded auditory input results from the intersection of classroom acoustics and poor teacher voice quality.



**3:15**

**3pAAc2. Speech intelligibility in auralized classrooms when the talker is wearing a face mask.** Pasquale Bottalico (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 South Sixth St., Champaign, IL 61820, pb81@illinois.edu), Silvia Murgia (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL), Giuseppina Emma Puglisi, Arianna Astolfi (Dipartimento di Energia, Politecnico di Torino, Torino, Italy), and Karen I. Kirk (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

This study explored the effects of wearing face masks on classroom communication. We evaluated the effects of three different types of face masks (fabric, surgical and N95 masks) on speech intelligibility presented to college students in auralized classrooms. To simulate realistic classroom conditions, speech stimuli were presented in the presence of speech-shaped noise with a signal-to-noise ratio of +3 dB under two different reverberation times (0.4 s and 3.1 s). The use fabric masks yielded significantly greater reduction in speech intelligibility compared to the other masks. Therefore, surgical masks or N95 masks are strongly recommended in teaching environments.

**3:35**

**3pAAc3. Higher speech levels in K-12 classrooms correlate with lower math achievement scores.** Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska—Lincoln, 1110 S 67th St., Omaha, NE 68182-0816, lilywang@unl.edu), Laura C. Brill (Threshold Acoust., Chicago, IL), Jayden Nord, and James Bovaird (Educational Psychol., Univ. of Nebraska—Lincoln, Lincoln, NE)

Sound levels in 275 K-12 midwestern classrooms have been logged every 10 s over two occupied school days (220 rooms over the 2015–2016 and 2016–2017 academic years) or four occupied school days (55 rooms in the 2017–2018 academic year). Measurements were made two or three times to capture data during both heating and cooling seasons. K-means clustering was used to group the data into times when speech was or was not occurring; then acoustic metrics were calculated from the clustered data. Demographic data and achievement data in the form of percentile ranks on math and reading tests were also collected for the students in each classroom and aggregated into classroom averages. Multivariate linear regression analysis on the initial dataset of 220 classrooms indicates that higher speech levels in classrooms correlate with lower math scores, with a significant interaction with the percentage of students receiving free or reduced-price lunches in the classroom. A statistically significant interaction is also found of non-speech levels and the percentage of gifted students in the classroom on reading scores. Data from the latter 55 classrooms are used to cross-validate the initial model. [Work supported by the United States Environmental Protection Agency Grant No. R835633.]

**3:55**

**3pAAc4. Psychoacoustics and classroom acoustics: New possibilities for interaction.** Pavel Zahorik (Dept. of Otolaryngol. & Comm. Dis., Univ. of Louisville, Louisville, KY 40292, pavel.zahorik@louisville.edu)

For more than a century, psychoacoustics has contributed to our understanding of how humans perceive sound and how the human auditory system encodes sound. Although general knowledge from psychoacoustics is clearly relevant to classroom acoustics, relatively little research has directly linked these two areas of study. Perhaps the closest link is in the detection and perception of a single ideal acoustic reflection, where psychoacoustics has documented effects of various factors, such as reflection delay, reflection direction, and source material. Of course, real rooms are vastly more complex acoustically, and thus, the psychoacoustic complexities already evident with single reflections surely multiply in real rooms. Historically, there have also been challenges in the experimental control of relevant room acoustic parameters. Virtual acoustics has largely solved this problem, but detailed scientific study of the perceptual aspects of room acoustics still involves exploring a very large parameter space, which can be prohibitively time-intensive for human subjects testing. Here, a computational modeling approach based on psychoacoustic principles is described that greatly facilities parameter space exploration. The modeling is applied to the problem of determining reverberation detection threshold, which provides critical and missing basic information regarding the limits of human sensitivity to room reverberation.

WEDNESDAY AFTERNOON, 9 DECEMBER 2020 — 1:05 P.M. TO 2:35 P.M.

# Session 3pAB

## Animal Bioacoustics: Parameters and Features of Animal Bioacoustic Signals

Marie J. Zahn, Chair
*School of Aquatic and Fishery Sciences, University of Washington, 1122 NE Boat St., Seattle, WA 98105*

**Chair's Introduction—1:05**

### *Contributed Papers*

**1:10**

**3pAB1. Song and hearing in different canary strains (*Serinus canarius*).** Robert J. Dooling (Dept. of Psych., Univ of Maryland, College Park, MD 20742, rdooling@umd.edu), Jane Brown (Psych., Univ of Maryland, Memphis, TN), Beth Brittan-Powell, Greg Ball (Psych., Univ of Maryland, College Park, MD), Matthew Conte, Karen Carleton (Biology, Univ. of Maryland, College Park, MD), and Farrah Madison (Psych., Univ. of Maryland, College Park, MD)

Canaries have been selectively bred for specific song characteristics (song canaries) or for morphology or plumage (type canaries) for centuries. Type canaries (e.g., Border and Gloster strains) retain song characteristics that are quite similar to those of wild canaries. By contrast, song canaries (e.g., Belgian Waterslager and Roller strains) have been selected for song types pleasing to the human ear, resulting in songs that, in most cases, are less complex, lower-pitched, and narrower in a frequency range than songs from wild canaries. We now suspect that song selection in the Belgian Waterslager song canary has either directly or indirectly resulted in high-frequency hearing loss associated with hair cell abnormalities. Here, we compare hearing in the Belgian Waterslager and several other type and song canaries including the American Singer Canary. Though bred only since the 1930s, American Singer canaries may also have a high-frequency hearing loss that looks very similar to that of the Belgian Waterslager and may involve similar pathologies. Illumina whole-genome sequencing has preliminarily identified a number of high-impact SnpEff variants in Belgian Waterslager and American Singer Canaries, some of which are related to deafness genes in mammals.

**1:30**

**3pAB2. Source level measurements of killer whale (*Orcinus orca*) echolocation clicks.** Jennifer Wladichuk (Univ. of Victoria/JASCO Appl. Sci., 3393 Robson Pl., Victoria, BC V9C 0J2, Canada, jwladichuk@uvic.ca), Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), Jens Koblitz (Max Planck Inst. of Animal Behavior, Constance, Germany), and Jack Lawson (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

Biosonar is vital to odontocetes (toothed whales, dolphins, and porpoises) for navigation and finding food. Consequently, numerous details of their echolocation click parameters, such as source level and beam width,

are known for many species; however, there is a data gap for killer whales (Orcinus orca). This information is essential for examining masking potential which could have an impact on their foraging abilities. The Southern Resident killer whales are an endangered population of orcas living along the Pacific coast of North America and underwater noise has been identified as a major threat towards their survival. This study used a 3D 24-element hydrophone array deployed in close proximity to wild killer whales to estimate source levels (SLs) and spectra of echolocation clicks. Future work will use these SLs to investigate masking potential from current and other noise scenarios.

**1:50**

**3pAB3. Vertical sonar beam width of wild belugas (*Delphinapterus leucas*) in West Greenland.** Marie J. Zahn (School of Aquatic and Fishery Sci., Univ. of Washington, 1122 NE Boat St., Seattle, WA 98105, mzahn@uw.edu), Jens Koblitz (Bioacoustics Network, Constance, Germany), and Kristin L. Laidre (Polar Sci. Ctr., Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Echolocation signals of wild beluga whales (*Delphinapterus leucas*) were recorded in 2013 using a vertical, linear 16-hydrophone array at two locations in the pack ice of Baffin Bay, West Greenland. Individual whales were localized for 12 min of 1:04 h of recordings where on-axis clicks were isolated to calculate sonar parameters. We report the first sonar beam estimate of *in situ* recordings of belugas with an average 3-dB asymmetrical vertical beam width of 5.7 deg, showing a wider ventral axis. This is narrower than the commonly used estimates in the literature obtained from captive whales, suggesting beluga sonar beam width may be different in captive and wild contexts and is not necessarily symmetrical. Apparent source levels ranged from 210 to 220 dB pp *re* 1 $\mu$Pa and whales were shown to vertically scan the array from 120 m distance. Our findings support the hypothesis that highly directional sonar beams and high source levels are an evolutionary adaptation for Arctic odontocetes to reduce noise and surface echoes from ice. These results provide the first baseline beluga sonar metrics from free-ranging animals using a hydrophone array and are important for acoustic programs throughout the Arctic, particularly for acoustic classification between belugas and narwhals (*Monodon monoceros*).

**2:10**

**3pAB4. Source levels and propagation modelling of whistles from a free-ranging rough-toothed dolphin.** Lis Bittencourt (MAQUA, Rio de Janeiro State Univ., São Francisco Xavier St., 524 - Maracanã, Rio de Janeiro 20550-900, Brazil, lis.bitt@gmail.com), Mariana Barbosa, Elitieri Santos-Neto, Tatiana Bisi, José Lailson-Brito, and Alexandre Azevedo (MAQUA, Rio de Janeiro State Univ., Rio de Janeiro, Brazil)

A lone adult individual of rough-toothed-dolphin had its acoustic behavior recorded with a calibrated system (Fostex FR22: 192 kHz sampling rate; C54 hydrophone: -165.0 dBV, 0.009 Hz–100 kHz). Moments when the animal approached and faced the hydrophone were noted. Whistles were counted in Raven 1.6 (Hann window, 512, 50% overlap). Duration (ms), minimum, maximum and peak frequencies (kHz), number of steps and step frequency (kHz) were extracted of whistles emitted when the animal was in a 1-m radius of the hydrophone. Source levels (SL) were estimated through power spectral density calculation in Matlab. SL at frequency parameters and at each whistle step were extracted and used in bellhop propagation models of a coastal scenario. 63 whistles were emitted within 1m. Frequencies ranged from 1.2 to 9.7 kHz and duration from 115.3 to 825.6 ms. Frequency and SL increased with step number, with 4.5 steps per whistle. Minimum frequencies ($1.9 \pm 0.6$ kHz) had smaller SL ($125.2 \pm 4.2$ dB *re* 1 $\mu$Pa) and decreased to 100 dB before 500 m. Peak frequencies ($6.7 \pm 1.0$ kHz) had mean SL of $149.5 \pm 8.5$ dB *re* 1 $\mu$Pa and reached 2500 m with 100 dB. Higher frequencies reached longer distances and depths.

# Session 3pAO

## Acoustical Oceanography: Munk Award Lecture

Andone C. Lavery, Cochair
*Woods Hole Oceanographic Institution, Woods Hole, MA 02543*

Grant B. Deane, Cochair
*Scripps Institute Of Oceanography, Code 0238, UCSD, La Jolla, CA 92093-0238*

**Chair's Introduction—2:50**

### *Invited Paper*

**2:55**

**3pAO1. Can we map the entire global ocean seafloor by 2030?** Larry A. Mayer (Ctr. of Coastal and Ocean Mapping, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, larry@ccom.unh.edu)

Recognizing the poor state of knowledge of ocean depths and the critical role such knowledge plays in understanding and maintaining our planet, the Nippon Foundation challenged the mapping community to produce a complete map of the world ocean seafloor by 2030. The resulting project, Seabed 2030, has already increased publicly-available data holdings of modern multibeam sonar data from 9% at the start of the effort in 2018 to 19% in the latest release of the GEBCO bathymetric compilation. Much of this initial increase has come through discovery of existing data. The challenge now is to complete new mapping (at resolution much higher than that achievable by satellite-altimetry derived bathymetry), an effort estimated to require approximately 200 ship-years (at a cost of $3-5B) using current technologies. Meeting this challenge will require innovative new technologies that can increase efficiency, cost-effectiveness and, capabilities. Autonomous vehicles can deliver standard mapping systems without the significant cost of crews and wind-powered autonomous systems, without the cost of crews or fuel. The real challenge however, is an acoustic one. Are new approaches to deep-sea mapping feasible that will represent a quantum leap in technology and make the aspirational goal of Seabed 2030 a reality?

## Session 3pBAa

# Biomedical Acoustics, Physical Acoustics, Structural Acoustics and Vibration, and Computational Acoustics: Fractional Calculus Models of Compressional and Shear Waves for Medical Ultrasound III

Sverre Holm, Chair
*Physics, University of Oslo, P. O. Box 1048, Blindern, Oslo N 0316, Norway*

**Chair's Introduction—1:05**

### *Contributed Paper*

**1:10**

**3pBAa1. Quasi-continuum theories of compressional wave propagation predicting amplitude-dependent anomalous frequency power law of attenuation.** 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) and Charles W. Holland (Portland State Univ., State College, PA)

A common assertion is that rational linear continuum mechanical models are incompatible with an attenuation exponent varying as first power of frequency at low frequencies, and that the attenuation in this limit always goes as frequency squared. Recently, the authors' attention was drawn to a paper [*J. Geophys. Res.* (1960)] by Leon Knopoff and Gordon MacDonald (K-M). They proposed a not-quite-linear one-dimensional wave equation that, with some math given in present paper, predicts attenuation proportional to frequency. An extension of the model to three dimensions begins with the Euler equation model that force on a fluid particle is proportional to negative of pressure gradient. The proportionality, however, depends on whether the particle is moving with or against the gradient. The increment is taken as a small constant times the "sign" function of the dot product of particle and pressure gradient vectors. With a perturbation technique similar to that of Krylov and Bogoliubov, one does indeed predict the linear frequency dependence. It is argued, however, that the abrupt discontinuity caused by the sign function is inconsistent with the notion of a rational linear continuum model. One can replace the sign function by a smooth function with desired asymptotic behavior, but the model then becomes nonlinear. This observation leads to the speculation that linear frequency dependence of attenuation is a nonlinear phenomenon that might not be observed if the amplitude is sufficiently low. [Work supported by ONR.]



### *Invited Paper*

**1:30**

**3pBAa2. Limitations on nonlinear fractional calculus models used in biomedical acoustics.** Blake E. Simon (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), John M. Cormack (Ctr. for Ultrasound Molecular Imaging and Therapeutics, Dept. of Medicine, Univ. of Pittsburgh, Pittsburgh, PA), and Mark F. Hamilton (Appl. Res. Labs., Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX 78712-1591, hamilton@mail.utexas.edu)

Fractional calculus models used for biomedical ultrasound are associated with attenuation proportional to $\omega^y$, where $y$ is typically in the range $1 < y < 2$. To determine whether the attenuation and accompanying dispersion are sufficient to stabilize shock formation, the models are formulated as a Burgers equation with the traditional loss term replaced by a fractional derivative of order $y$. For $y < 1$ the resulting equation predicts unphysical solutions beyond the shock-formation distance. The second example pertains to nonlinear Lucassen interface waves, a model equation for which has been proposed to describe mechanical perturbations that accompany the transmission of nerve impulses. Linear Lucassen waves are defined by a second-order space derivative and a fractional time derivative of order 3/2, which falls between order 2 in the wave equation and order 1 in the diffusion equation. The resulting attenuation is proportional to $\omega^{3/4}$, and the corresponding nonlinear "fractional diffusive waves," while strongly attenuated on the scale of a wavelength, may be lacking essential physics beyond the predicted shock-formation distance. Calculations are presented that determine wave amplitudes and propagation distances for which these two fractional calculus models may be of questionable physical significance due to nonlinearity.
[B.E.S. is supported by the ARL:UT McKinney Fellowship in Acoustics.]

## *Contributed Paper*

**1:50**

**3pBAa3. Nonlinear Lucassen waves with interface viscosity.** Blake E. Simon (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, blakesimon8@utexas.edu) and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Linear theory for quasi-longitudinal surface wave propagation along an elastic interface coupled to a viscous, incompressible liquid was first developed by Lucassen [*Trans. Faraday Soc*. 1968]. Lucassen waves are modeled with a fractional diffusion-wave equation in which the order of the fractional time derivative is 3/2. Nonlinearity in the elastic interface was taken into account recently by Kappler *et al*. [*Phys. Rev. Fluids* 2017]. Nonlinear Lucassen interface waves exhibit certain features associated with the mechanical disturbance that accompanies the electric action potential in the biological membranes of nerve axons, such as the "all-or-none" principle in which wave speed and pulse shape change dramatically above some amplitude threshold. While Lucassen waves are highly damped, for nonlinear propagation the attenuation described by the fractional time derivative provides insufficient energy loss near regions where shocks form in the waveform, resulting in the failure of conventional numerical algorithms such as Runge-Kutta schemes. Presented here is a modified model equation for nonlinear Lucassen waves that includes viscoelastic effects in the interface. The inclusion of viscosity in the interface results in greater losses at shock fronts and increased stability for numerical calculations. [B.E.S. is supported by the ARL:UT Chester M. McKinney Graduate Fellowship in Acoustics.]

WEDNESDAY AFTERNOON, 9 DECEMBER 2020 — 1:05 P.M. TO 2:15 P.M.

# Session 3pBAb

# Biomedical Acoustics: General Biomedical Acoustics: Transcranial Focused Ultrasound

Jonathan A. Kopechek, Cochair
*Bioengineering, University of Louisville, 2301 S. Third St., Paul C. Lutz Hall, Room 419, Louisville, KY 40292-0001*

**Chair's Introduction—1:05**

## *Contributed Papers*

**1:10**

**3pBAb1. Characterize brain tumor genetic signatures with focused ultrasound-enabled liquid biopsy.** Christopher P. Pacia (Dept. of Biomedical Eng., Washington Univ. in St. Louis, Saint Louis, MO, cpacia@wustl.edu), Lifei Zhu, Jinyun Yuan, Yimei Yue (Dept. of Biomedical Eng., Washington Univ. in St. Louis, Saint Louis, MO), and Hong Chen (Washington Univ. in St. Louis, St. Louis, MO)

Although blood-based liquid biopsy is a promising noninvasive diagnostic technique, the blood-brain barrier (BBB) hinders the efficient transport of brain tumor biomarkers into the bloodstream. In our previous studies, we demonstrated the feasibility for focused ultrasound-enabled liquid biopsy (FUS-LBx) to increase BBB permeability and enhance the transport of glioblastoma (GBM)-specific biomarkers into the bloodstream of mice using mRNA as a model biomarker. The objective of this study was to demonstrate the clinical translatability of FUS-LBx by evaluating the enhanced transport of the GBM-specific gene mutation, epidermal growth factor receptor variant III (EGFRvIII). Rats were randomly assigned to the treated ($n = 10$) or control groups ($n = 6$) 10 days after F98-EGFRvIII tumor cell implantation. MRI scans guided the combined FUS and microbubble sonication to the tumor center and verified successful BBB disruption. Blood was collected after FUS sonication or sham treatment. Droplet digital polymerase chain reaction was used to analyze the EGFRvIII complementary DNA. Rat brains were collected for histological analysis of potential FUS-induced tissue damage. MRI scans confirmed successful FUS-induced BBB opening. The EGFRvIII mutation that was undetectable by conventional LBx (no FUS), was detected by FUS-LBx. Histological analysis showed no significant tissue damage caused by FUS. This study demonstrated that FUS-LBx has the clinical translational potential for the noninvasive characterization of the GBM genetic profiles.

**1:30**

**3pBAb2. Segmentation of induced substantia nigra from transcranial ultrasound images using deep convolutional neural network.** Niranjan Thirusangu (School of Elec. Eng. and Comput. Sci., Penn State Univ., W136 Westgate Bldg., University Park, PA 16802, niranjanthirusangu@gmail.com), Thyagarajan Subramanian (Dept. of Neurology Neural - Dept. of Behavioral Sci., Penn State College of Medicine, Hershey, PA), and Mohamed Almekkawy (School of Elec. Eng. and Comput. Sci., Penn State Univ., State College, PA)

Parkinson's disease (PD) is an age-related neurodegenerative disorder, whose early diagnosis is challenging. PD neuropathology is characterized by a selective loss of dopaminergic neurons in the substantia nigra (SN). The echogenicity of SN is considered as an important biomarker for diagnosing PD. Since Ultrasound is well suited for measuring echogenicity, transcranial ultrasound images (TCUI) are used to diagnose PD and have become an industrial standard. But, ultrasound images usually have low resolution and are noisy compared to other medical imaging modalities. Thus, this whole method relies on the experience of the clinician to identify SN from the TCUI. To automate the process, we propose a deep convolutional neural network based on the U-Net architecture with a weighted binary

cross-entropy (WBCE) loss function to do semantic segmentation of SN from the TCUI obtained. To alleviate the negative effects caused by noisy images, third-order Volterra filter is used for pre-processing. Furthermore, we compare the convergence rate of WBCE with standard BCE and other standard loss metrics, where WBCE outperforms others. We then, compare the results of the proposed U-Net architecture with a DenseNet based architecture and a U-Net and DenseNet based hybrid architecture, where we obtain better accuracy with the proposed architecture.

**1:50**

**3pBAb3. Polymers in high-intensity focused ultrasound fields.** Kaiyuan Peng (Mech. Eng., Virginia Tech, 635 Prices Fork Rd., Blacksburg, VA 24061, kaiyuan@vt.edu), Shima Shahab, and Reza Mirzaeifar (Mech. Eng., Virginia Tech, Blacksburg, VA)

High-intensity focused ultrasound (HIFU) is a promising stimulus that has received extensive attention during the past two decades due to numerous advantages of this technology when compared to the conventional methods. For HIFU-responsive polymers, their applications could be extended to controlled drug delivery, soft robotics, and flexible electronics. In this work, experiments show that polymers heat up in a different manner when they are subjected to HIFU, compared to the case they are subjected to heat sources directly. However, the origins of this difference are still entirely unknown. To further investigate, molecular dynamics (MD) simulations are performed to study the thermal effect of polymers induced by HIFU. We found the difference in the macroscale response of polymers subjected to HIFU strongly depends on the polymer chains, composition, and structure. The frequency-dependent viscoelasticity, measured by stress-strain phase lag, the reputation motion of the chains, and the vibration-induced local mobility, quantified by the root mean square fluctuation (RMSF) contribute to the observed difference in the HIFU-induced thermal effects. The simulation results show a good accordance with the experiment qualitatively, and provide the fundamental mechanism for designing and optimizing stimuli-responsively polymeric devices. [Work supported by: this work was supported by NSF Grant No. CMMI-2016474.]

WEDNESDAY AFTERNOON, 9 DECEMBER 2020 — 1:05 P.M. TO 2:15 P.M.

## Session 3pCA

## Computational Acoustics, Underwater Acoustics, Structural Acoustics and Vibration, and Acoustical Oceanography: Acoustic Optimization: Methods and Applications III

Micah Shepherd, Chair
*The Pennsylvania State University, PO Box 30, State College, PA 16801*

**Chair's Introduction—1:05**

3p WED. PM

### *Contributed Papers*

**1:10**

**3pCA1. Adjoint-based optimization to minimize scattered pressure fields.** Benjamin C. Treweek (Computational Solid Mech. & Structural Dynam., Sandia National Labs., P.O. Box 5800, Albuquerque, NM 87185, btrewee@sandia.gov), Timothy Walsh (Simulation Modeling Sci., Sandia National Labs., Albuquerque, NM), Clay M. Sanders, Wilkins Aquino (Dept. of Civil and Environ. Eng., Duke Univ., Durham, NC), Samuel D. Parker, and Michael R. Haberman (Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

Acoustic and elastic metamaterials have shown potential for many different applications in wave manipulation and structural isolation. Minimization of a scattered pressure field has been identified as an important use for such materials, and a wide variety of designs have been proposed for this purpose. Several of these designs are difficult to manufacture (e.g., pentamode designs), and many others involve numerous tunable parameters and a large design parameter space (e.g., distributions of resonators). This presents a challenge for global search-based optimization and raises the need for gradient-based methods. In this work, adjoint-based, PDE-constrained optimization is used to improve the feasibility of minimizing a scattered pressure field via an optimal set of channels in an annulus surrounding a rigid scatterer. Two different optimization strategies are investigated: material identification, where the physical properties of each channel are treated as unknown, and force identification, where the magnitude and phase of the reflected field is treated as unknown at various patches on the annulus. Results are presented for both a single plane wave direction and multiple plane wave directions, as well as for both a single frequency and multiple frequencies. [S.N.L. is managed and operated by NTESS under DOE NNSA Contract No. DE-NA0003525.]

**1:30**

**3pCA2. Investigation of optimization techniques on structural-acoustical shaped concrete slabs in buildings.** Jonathan Broyles (Architectural Eng., The Penn State Univ., 510 Toftrees Ave. Apt # 331, State College, PA 16803, jmb1134@psu.edu), Micah Shepherd (The Penn State Univ., State College, PA), and Nathan C. Brown (Architectural Eng., The Penn State Univ., University Park, PA)

Computational tools have become integrated into design practice at the building scale and component scale. While there has been a resurgence of optimization techniques in room acoustics, research has been limited on utilizing optimization techniques on building components, such as a building's structural floor. Quality design solutions must be found at the component scale to accommodate for increased urbanization, environmental concerns,

building utilization, and the well-being of the occupants, especially in relation to the tenants' acoustic environment. This presentation will discuss the use of several design space exploration and optimization approaches to generate and consider multiple permutations of shaped concrete floor designs. The shape of the ribbed slab will be varied in order to improve both the embodied energy of the concrete slab, which is proportional to mass, as well as the sound transmission class (STC). Three computational techniques (Latin Hypercube Sampling, multi-objective evolutionary optimization and constrained optimization) will be used to determine trade-offs in the design. The advantages and disadvantages of each technique will be highlighted with respect to the trade-off between reduced mass and improved STC. Finally, the importance of model resolution will be discussed in early design space exploration and optimization procedures.

**1:50**

**3pCA3. Testing and optimization of 3D-printed volumetric diffusor arrays for attenuation of blast noise.** Gordon M. Ochi (Construction Eng. Res. Lab., US Army ERDC, 2902 Newmark Dr., Champaign, IL 61822, Gordon.M.Ochi@erdc.dren.mil), Kyle Dunn, Michael B. Muhlestein (Cold Regions Res. and Eng. Lab., U.S. Army Eng. Res. and Development Ctr., Hanover, NH), Tanner Wood, Megan Kreiger, and Michelle E. Swearingen (Construction Eng. Res. Lab., US Army ERDC, Champaign, IL)

3D-printed volumetric diffusing array structures, composed of many arbitrarily shaped pillars in an optimized pattern, have previously been shown to be very effective for blast noise attenuation in a scale model experiment (Ochi *et al.*, 2019). This work expands upon previous progress by investigating innovative optimization schemes for the array designs, and further testing these optimized arrays in scale model experiments with a spark gap generator operating as a surrogate blast noise source. Properties of the propagated signal, including directivity, duration, and spectral characteristics, are discussed, as well as how these properties depend on the choice of volumetric diffusor. Finally, the propagated waveform, spectra, and effective attenuation predicted using 2D-FDTD simulations are compared with the results of the experiment as a method of testing the validity of the simulations.

WEDNESDAY AFTERNOON, 9 DECEMBER 2020 1:05 P.M. TO 2:35 P.M.

## Session 3pEAa

## Engineering Acoustics and Architectural Acoustics: Microphones: From Rock Stars to Rockets III

Vahid Naderyan, Cochair
*University of Mississippi, 1151 Maplewood Drive, Itasca, IL 60143*

Sandra J. Guzman, Cochair
*Shure, Inc., 4800 W Touhy Ave., Niles, IL 60714*

Edward M. Okorn, Cochair
*GRAS NA, Inc., 2234 East Enterprise Parkway, Twinsburg, OH 44087*

Neil A. Shaw, Cochair
*Menlo Scientific, PO Box 1610, Topanga, CA 90290*

**Chair's Introduction—1:05**

### *Invited Papers*

**1:10**

**3pEAa1. MEMS microphones in commercial applications and beyond: Technology trends in the next decade.** Michael Pedersen (Knowles LLC, 1151 Maplewood Dr., Itasca, IL 60143, michael.pedersen@knowles.com)

Since the initial introduction in the early 2000s, MEMS-based microphones have grown to become the preferred choice in a large range of commercial applications. This growth has been driven by continuous improvements in transduction performance, product consistency, and device ruggedness to meet ever more stringent application requirements. The emergence and rapid development/growth of small wearable devices, has given rise to a new set of possibilities and requirements for the acoustic transducer. In these applications power consumption, size, and ingress immunity are of major importance. While existing MEMS microphone topologies are being adjusted and adopted for this purpose, there are also opportunities for different structures and new operational schemes to further enhance product performance. In the area of acoustic performance, MEMS microphone technology has reached some fundamental barriers, particularly on size, at which entirely new approaches will be necessary to change the performance/size equation. In this presentation, a review will be given of MEMS microphone technology and how it got us to where we are today. In addition, projections will be shared on where the technology might go in the future where fully sealed, high performance, micro-scale devices are imaginable.

**1:30**

**3pEAa2. Engineering an optical microphone for consumer electronics.** Neal A. Hall (ECE, Univ. of Texas, Austin, 10100 Burnet Rd., Bldg. 160 Rm. 1.108, Austin, TX 78702, nahall@mail.utexas.edu)

Between the years 2010–2015, a small start-up in Austin, TX was focused on commercializing an optical MEMS microphone to address high-volume consumer electronics applications. The motivation for the optical approach was to achieve a higher signal-to-noise ratio (SNR) than capacitive MEMS microphones. Similar to other commercial MEMS microphone offerings, the product was required to be small size, relatively low-cost, and surface-mountable. This presentation will focus on the challenges faced in engineering an optical microphone to meet these requirements. The particular approach used a diffractive optical element fabricated directly into the backplate of an otherwise conventional microphone die. Vertical-cavity surface emitting lasers (VCSELs) resided inside the die's etched cavity. Completed pilot-scale samples proved capable of demonstrating high SNR, but ultimately the company made a pivot to focus engineering and business-development resources on a similar product line—ground-motion sensors and seismometers. The company has been manufacturing and selling seismometers 2015–present.

**1:50**

**3pEAa3. Ultrasonic anemometry on Mars.** Robert D. White (Mech. Eng., Tufts Univ., 200 College Ave., Medford, MA 02155, r.white@tufts.edu), Edward S. Schmid (Mech. Eng., Tufts Univ., Medford, MA), Ian Neeson (VN Instruments, Brockville, ON, Canada), Jonathan P. Merrison, Jens J. Iversen (Dept. of Phys. and Astronomy, Univ. of Aarhus, Aarhus, Denmark), and Don Banfield (Cornell Ctr. for Astrophysics and Planetary Sci., Cornell Univ., Ithaca, NY)

On Mars, we have yet to fully quantify and understand atmospheric turbulent transport (Banfield, *JASA*, 2016). The key required instrument is a wind sensor that can resolve horizontal and vertical winds, as well as the perturbations associated with turbulent eddies. On the surface, 10–20 Hz sampling and a sensitivity of 5 cm/s is required (McBean, 1972). Viking, Pathfinder, Curiosity, and Insight all used hot film or hot wire anemometers that can be confused by radiative heating and all have a response time of approximately 1 Hz. Sonic anemometry is an alternative high speed, high accuracy approach which uses differences in the acoustic phase delay to determine flow speed. Operation on Mars is challenging due to the low acoustic source strength and high attenuation in the thin (6–11 mbar) $CO_2$ atmosphere, as well as large temperature variations (-100 °C to 20 °C). In this paper, we describe a three axis continuous wave piezoelectric anemometer with a high resolution and high precision analog phase detection circuit. Testing was in dry carbon dioxide between 2 and 20 mbar near 20 °C. At 6 mbar, we demonstrate resolution better than 10 cm/s, and accuracy of 3% at flow speeds up to 12 m/s. Low temperature testing is ongoing.

**2:10**

**3pEAa4. Weird science microphones are coming.** Mike Klasco (Menlo Sci., 5161 Rain Cloud Dr., Richmond, CA 94803, mike@menloscientific.com) and Neil A. Shaw (Menlo Sci., Topanga, CA)

Electret condenser microphone elements ruled for decades, are being displaced, at first in the late 1990s, and more rapidly in the last decade, by capacitive MEMs. Breaking the signal to noise ceiling at 70 dB in small consumer electronics microphone formats has been a tenacious barrier for far field voice control and sound pickup. Some existing conventional solutions and those just below the radar are more than promising. The advances in materials science in the last few years have been remarkable. Product demand in all categories, especially mobile electronics and voice command, are driving this rapid advancement. Some of these developments in materials and their application to existing microphones and for the next generation microphone products will be revealed. "Weird microphone science projects" that are jelling toward commercial productization, such as graphene electrostatic and ribbon mics, optical microphones, piezo MEMs, insect eye mics, and more will be explored.

# Session 3pEAb

## Engineering Acoustics and Architectural Acoustics: Microphones: From Rock Stars to Rockets IV

Vahid Naderyan, Cochair
*University of Mississippi, 1151 Maplewood Drive, Itasca, IL 60143*

Sandra J. Guzman, Cochair
*Shure, Inc., 4800 W Touhy Ave., Niles, IL 60714*

Edward M. Okorn, Cochair
*GRAS NA, Inc., 2234 East Enterprise Parkway, Twinsburg, OH 44087*

Neil A. Shaw, Cochair
*Menlo Scientific, P O Box 1610, Topanga, CA 90290*

**Chair's Introduction—2:50**

### *Contributed Papers*

**2:55**

**3pEAb1. Cryogenic testing of MEMS microphones towards their utilization as a quench detection method in superconducting magnets.** Robert D. White (Mech. Eng., Tufts Univ., 200 College Ave., Medford, MA 02155, r.white@tufts.edu), Zijia Zhao, Casey Owen, Mischael Anilus (Mech. Eng., Tufts Univ., Medford, MA), Steve Chau, Amish Desai, Michael Emerling (Tanner Res., Inc., Monrovia, CA), Luisa Chiesa (Mech. Eng., Tufts Univ., Medford, MA), and Makoto Takayasu (Plasma Sci. and Fusion Ctr., Massachusetts Inst. of Technol., Cambridge, MA)

High Temperature Superconducting (HTS) tapes such as Rare Earth Barium Copper Oxide (REBCO) are an attractive option for high field magnets operated in cryogenic fluids. However, quench events can occur in which the conductor locally loses its superconducting properties. It is critical to rapidly detect and respond to such an event. Conventional quench detection methods using voltage taps are difficult in HTS devices, since the normal zone propagation velocities are 2–3 orders of magnitude lower in HTS compared to low temperature superconductors (Iwasa, *Cryogenics*, 2003). We propose an alternative in which a linear array of MEMS microphones is distributed down the central cooling channel of a cable in-conduit conductor. The array can detect the acoustic signature caused by a quench event which propagates in the cooling fluid. The proposed method differs from Acoustic Emission (AE) detection, which uses sensors mounted on the magnet surface to detect structural vibration (Tsukamoto, *Appl. Phys. Lett.*, 1981). In order to implement the system, MEMS microphones and preamplifiers must operate in cryogenic fluids. We report on characterization of commercial MEMS microphones in cryogenic gaseous helium between 0.5 and 1.5 bar down to 20 K, and in liquid nitrogen at 1 bar and 77 K.

**3:15**

**3pEAb2. Analytical and computational solutions for the acoustic damping of perforated microstructures.** Vahid Naderyan (Univ. of MS, 1151 Maplewood Dr., Itasca, IL 60143, vahid.nad@gmail.com), Richard Raspet, and Craig J. Hickey (Univ. of MS, University, MS)

Efficient models for the viscothermal acoustic damping of perforated micro-electro-mechanical systems (MEMS) are essential in the designs of MEMS devices. In this work, the low reduced-frequency (LRF) method for viscothermal acoustic propagation is utilized to develop an analytical model for perforated MEMS. Also, a 3D finite element method (FEM) model including viscous and thermal effects is developed for the numerical computation of the problem. The results of the proposed analytical solution are in good agreement with the FEM. The applications of the model to predict and also to optimize the performance of the MEMS microphones will be discussed.

**3:35**

**3pEAb3. Microphone length and its effect on vibration interference.** Jonathan D. Walsh (Mech. Eng. Dept, Binghamton Univ., 85 Murray Hill Rd., Binghamton, NY 13902, jwalsh3@binghamton.edu) and Ronald Miles (Eng., Binghamton Univ., Binghamton, NY)

One-dimensional duct models yield the deflection of an idealized massless diaphragm subjected to pressure and vibration. Additional ANSYS models also include the effects of a flexible, thin diaphragm. These models are used to predict the acceleration pressure that a microphone senses when subject to vibration, in terms of the effective mass added by the loading due to nearby air. This effective mass value is shown to be proportional to one half the length of the back volume, plus the length of the front volume. Previously measured data from electrets and MEMS microphones are shown to support this conclusion. The length-dependence of the vibration pressure suggests that a shorter microphone with a diaphragm placed as close as possible to the outside air will have minimal vibration pressure sensitivity. [This work is supported by a grant from the NIH National Institute on Deafness and other Communication Disorders (1R01DC017720-01).]

## Session 3pED

## Education in Acoustics and Women in Acoustics: Hands-On Demonstrations: Acoustics at Home

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

Keeta Jones, Cochair
*Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300*

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 student viewrs to try at home. 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. An experienced acoustician will be stationed at each demonstration who will help the students understand the principle being illustrated. Any acousticians wanting to participate in this fun event should e-mail Keeta Jones (kjones@acousticalsociety.org).

WEDNESDAY AFTERNOON, 9 DECEMBER 2020 2:50 P.M. TO 4:00 P.M.

## Session 3pID

## Interdisciplinary: Hot Topics in Acoustics

Subha Maruvada, Chair
*U.S. Food and Drug Administration, 10903 New Hampshire Ave., Bldg. WO 62-2222, Silver Spring, MD 20993*



**Chair's Introduction—2:50**

### *Invited Papers*

**2:55**

**3pID1. Virtual acoustic environments for perception research.** Michael Vorlaender (Inst. of Tech. Acoust., RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52056, Germany, mvo@akustik.rwth-aachen.de)

Human sound perception includes the ability to process auditory information from dynamic acoustic situations, which is an essential component for communication and orientation in everyday life. In contrast, standard psychoacoustic experiments don't include real-life situations.They include rather simple acoustic stimuli with nominally high reproducibility but low significance for situations in real life. The gap in research on auditory perception between the laboratory and real life can be closed now. Virtual Reality technology is a very powerful tool which connects real-time computer simulation with 3D devices for auditory-visual impressions. Virtual acoustics can incorporate any situational context and scene conditions with sources and sound propagation features in given environments. In consequence, ground-breaking research in auditory perception can be expected with higher degree of realism and with still high reproducibility. The presentation will highlight guidelines for creation of Virtual Acoustic Environments and examples for application in noise assessment, psychoacoustic research, and hearing diagnosis and rehabilitation.

**3:15**

**3pID2. Additive manufacturing in musical acoustics.** John Granzow (Performing Arts Technol., Univ. of Michigan, 1100 Baits Dr., Ann Arbor, MI 48109, jgranzow@umich.edu)

Additive Manufacturing (AM) affords the production of complex geometries for acoustic research. The rapid materialization of parametric models facilitates the practical comparison of related sounding or sound-filtering objects. This accelerated fabrication cycle continues our field's tradition of establishing and demonstrating results through making, with applications in structural acoustics (metamaterials), architectural acoustics (diffuser design) and musical acoustics (instrument design). This talk focuses on applications of AM in musical acoustics: the replication of known instrument geometries, the precise transformation of those geometries in order to test predictions, and explorations of the musicality of neighboring forms. Results, limitations and future prospects are considered.

**3:35**

**3pID3. Flipping the lab: Portable automated rapid testing (PART) systems for psychological acoustics.** Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., 3181 Sam Jackson Park Rd., Portland, OR 97239, gallunf@ohsu.edu)

The second decade of the 21st Century has seen the rise of consumer technology capable of producing auditory signals at a level of precision only available to laboratory scientists and engineers just a few years earlier. This has allowed researchers to "flip the lab" and begin creating testing systems that can be taken to the participants rather than bringing the participants to the systems. This talk will describe how freely available tablet and smart-phone applications, low-cost portable hearing-aid development systems, and programmable virtual reality systems are all poised to change that way that research is conducted. Discussion will include the research that has been done validating the existing approaches and describing those guides to best practices that exist (or need to). Finally, this talk will discuss the ways that such approaches are essential aspects of lowering barriers to who can be involved in clinical research, both as experimenters and as participants.

WEDNESDAY AFTERNOON, 9 DECEMBER 2020 — 1:05 P.M. TO 2:35 P.M.

## Session 3pMUa

## Musical Acoustics: General Topics in Musical Acoustics II

Bozena Kostek, Cochair
*Audio Acoustics Lab., Gdansk Univ. of Technology, Narutowicza 11/12, Gdansk, 80-233, Poland*

Tim Ziemer, Cochair
*Bremen Spatial Cognition Center, University of Bremen, Enrique-Schmidt-Str. 5, Bremen, 28359, Germany*

**Chair's Introduction—1:05**

### *Contributed Papers*

**1:10**

**3pMUa1. Determining low frequencies of the Japanese koto using a finite element model based on a computed tomography scan.** A. K. Coaldrake (Pan Pacific Technologies, Pan Pacific Technologies, Adelaide, South Australia 5075, Australia, coaldrake@panpacific.com.au) and Eric H. Dunlop (Pan Pacific Technologies, Adelaide, South Australia, Australia)

The Japanese koto is a plucked 13-string wooden instrument noted for its complex resonances. A finite element model of the koto has been developed based on a computed tomography (CT) scan and has previously been reported. It can now be used as a tool to undertake more detailed examination of the koto's sound behavior. While first iterations of the model were found to predict frequencies above 100 Hz, predictions for frequencies below 100 Hz were less accurate. Physical studies however have shown many low frequencies do occur and are not artefacts. This paper discusses initial simulations with the model and more recent studies with Gaussian pulses that reveal low frequencies not previously obtained from the CT model. Their contribution to the characteristics of the sound behavior of the koto are discussed.

**1:30**

**3pMUa2. Free vibration of a kalimba tine model beam with offset boundary condition.** Daniel Ludwigsen (Phys., Kettering Univ., 1700 University Ave., Flint, MI 48504, dludwigs@kettering.edu) and Jesus Huizar (Phys., Kettering Univ., Flint, MI)

A study was conducted to better understand the offset boundary condition of the kalimba; this study explored the free vibration of a beam with one end free and the other end subject to an offset clamped boundary condition. The cantilever beam was resting on a table, which provided a solid boundary below the beam. The top surface of the beam was clamped at a variable distance from the table edge. The excitation was a constant initial displacement of the free end, after which the entire beam was free to vibrate. The spectra of beam vibration and the sound produced showed harmonic peaks at integer multiples of the fundamental frequency, in addition to mode frequencies expected with Euler-Bernoulli beam theory.

**1:50**

**3pMUa3. Quantitative analysis of the kowangan resonator in the bundengan musical instrument.** Gea O. Parikesit (Universitas Gadjah Mada, Jalan Grafika 2, Yogyakarta 55281, Indonesia, geaofp@yahoo.com) and Indraswari Kusumaningtyas (Dept. of Mech. and Industrial Eng., Universitas Gadjah Mada, Yogyakarta, Indonesia)

This paper presents a quantitative analysis of the kowangan, a shield-shaped resonator used in the bundengan, a bamboo-based musical instrument from Indonesia. The kowangan resonator is made from a woven lattice of bamboo splits, covered with layers of bamboo sheaths, and then secured with palm sugar fibers. To gain insight into how this resonator works, it is required to first determine the shape of the resonator. Previously, we have developed a computational model of the resonator shape, where the distribution of curvatures in the resonator surface is driven by the minimization of potential bending energy in the woven lattice of bamboo splits. In the present study, we report the latest development in this work, where we have designed and built an optical imaging setup to measure the actual shape of a real kowangan resonator. Our measurement data allows for further improvements on our computational model, and consequently a deeper insight into the physics of the kowangan. The ultimate aim of this study is to build a full virtual prototype, not only of the kowangan resonator, but also of the whole bundengan instrument, which will be useful for the conservation and documentation of this endangered cultural heritage.

**2:10**

**3pMUa4. Acoustical analysis of additively manufactured snare drums.** Ethan Switzer (Mech. Eng., Univ. of Hartford, West Hartford, CT) and Christopher M. Jasinski (Mech. Eng., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, jasinski@hartford.edu)

Additive manufacturing (3D printing) has seen tremendous growth over the past decade and is finding many fields of application, including in musical acoustics. Klapel Instruments has recently developed multiple snare drum prototypes using additive manufacturing production for the shell and rims of the drum. For the past two years, two pairs of undergraduates at the University of Hartford have worked with faculty to assess the acoustical properties of these drums, specifically focused on determining any acoustic differences caused by the new manufacturing technique and materials. Several traditionally manufactured drums along with the additively manufactured drums were tested structurally, performed on, audio recorded, and subjectively assessed. Modal, spectral, and subjective analysis along with a comparison between the drums created using additive manufacturing and the traditionally manufactured drums will be presented.

WEDNESDAY AFTERNOON, 9 DECEMBER 2020 — 2:50 P.M. TO 4:00 P.M.

## Session 3pMUb

## Musical Acoustics: General Topics in Musical Acoustics III

Bozena Kostek, Cochair
*Audio Acoustics Lab., Gdansk Univ. of Technology, Narutowicza 11/12, Gdansk, 80-233, Poland*

Tim Ziemer, Cochair
*Bremen Spatial Cognition Center, University of Bremen, Enrique-Schmidt-Str. 5, Bremen, 28359, Germany*

**Chair's Introduction—2:50**

### *Contributed Papers*

**2:55**

**3pMUb1. String instrument acoustic transfer processing.** Mark Rau (Music, Stanford Univ., 660 Lomita Court, Stanford, CA 94305, mrau@ccrma.stanford.edu), Jonathan S. Abel, Julius O. Smith (Music, Stanford Univ., Stanford, CA), and Doug L. James (Comput. Sci., Stanford Univ., Stanford, CA)

A real-time method of string instrument acoustic transfer which includes damping is proposed. Acoustic transfer of string instruments is relevant when trying to make a non-resonant instrument, such as an electric guitar, sound more similar to an acoustic guitar. Unlike previous acoustic transfer methods which only perform equalization, this method includes damping changes by using a time-varying filter which adds frequency-dependent exponential damping. Efficient digital filters are fit to bridge admittance measurements of an acoustic guitar and used to create equalization filters as well as damping correction filters. The damping correction filters work in real-time as they are triggered by onset and pitch detection of the signal measured through an under saddle pickup to determine the intensity of the damping.

3:15

**3pMUb2. Noninvasive methods for quantifying sound post placement in a cello.** Eric Rokni, Molly Smallcomb (Graduate Program in Acoust., Penn State Univ., State College, PA), Thomas Blanford (Appl. Res. Lab., Penn State Univ., PO Box 30, State College, PA 16804, teb217@arl.psu.edu), and Micah Shepherd (Appl. Res. Lab., Penn State Univ., State College, PA)

The position and orientation of the sound post in violin family instruments has long been known to color the sound of the instrument. Small changes in the position of the sound post relative to the top and back plates of the instrument can result in clearly audible changes in the instrument's tone. Quantifying the effect of the sound post's placement on tone requires precise localization of the sound post. Luthiers have developed simple techniques to localize the post relative to the top of the instrument. Precisely localizing the post relative to the back, however, typically requires disassembly of the endpin. This talk will explore nondestructive and noninvasive techniques to localize the sound post relative to the back of the instrument. Results of these techniques will be compared for two cellos – one with a laminate construction and the other carved. Finally, the effects of differences in material properties on these localization techniques will be discussed.

3:35

**3pMUb3. Quantifying the cumulative effects of sustained excitation on the vibrational response and radiated sound of violins: Preliminary results.** Kourtney Adkisson (Dept. of Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 989226-7422, Kourtney.Adkisson@cwu.edu) and Andrew A. Piacsek (Phys., Central Washington Univ., Ellensburg, WA)

There is a widely accepted belief among luthiers and performers that consistent playing of stringed instruments, especially in the violin and lute families, is associated with improved sound and playability. Because it is thought that a new instrument needs to be played for some time in order to achieve its performance potential, many luthiers "pre-vibrate" new instruments before they are sold or delivered to customers. Very few studies have been performed to investigate the effect of artificial stimulation of new stringed instruments and the phenomenon of "playing-in" is still poorly understood. A long-term project underway at Central Washington University seeks to measure changes in vibrational and acoustic response that can be attributed to sustained mechanical stimulus. Three new sibling violins (same model and month of manufacture) were recently acquired: one is a control, while the other two will be mechanically excited at the bridge with a broadband signal, each with a different amplitude, continuously for several months. Prior to initiating the sustained excitation, a thorough study was conducted of the variability and uncertainty in measurements of the vibrational and acoustic response of each violin; this will serve as a baseline for interpreting the significance of subsequent changes.

WEDNESDAY AFTERNOON, 9 DECEMBER 2020 — 1:05 P.M. TO 2:15 P.M.

## Session 3pNSa

## Noise, Architectural Acoustics, and Structural Acoustics and Vibration: In Memory of Richard Lyon III

Gregory C. Tocci, Cochair
*Cavanaugh Tocci Associates, Inc., 327F Boston Post Road, Sudbury, MA 01776*

Patricia Davies, Cochair
*Ray W. Herrick Laboratories, Purdue University, 177 S Russell Street, West Lafayette, IN 47907-2099*

**Chair's Introduction—1:05**

### *Invited Papers*

1:10

**3pNSa1. The inventiveness of Richard H. Lyon.** Ilene J. Busch-Vishniac (Sonavi Labs, 1100 Wicomico Rd., Ste. 600, Baltimore, MD 21230, ilene@sonavilabs.com)

In the course of his career, Dick Lyon considered a large number of noise topics, including transportation noise, noise propagation in urban environments, statistical energy analysis, machinery diagnostics, product sound quality, and the importance of phase in acoustic transfer functions. What ties the body of Dick's work in these areas together are two features: the sheer inventiveness displayed both in theoretical and experimental approaches, and the focus on applied acoustics problems. For instance, Dick and collaborator Richard Cann designed a novel spark sound source to use for scale modeling. Dick's work on SEA similarly reflected a willingness to plow new ground in order to find the tools able to provide answers to vexing acoustical challenges. The majority of Dick's work can be traced to his desire to improve products by modifying the sound produced or to use the sound to identify functioning issues within the product. Dick leaves behind a legacy of accomplishments in the product acoustics arena. On a personal note, Dick was fond of pushing himself to the limit, whether in academic endeavors or in personal accomplishments, such as his rowing. He loved cars and was a great mentor to a large number of accomplished acousticians.

1:30

**3pNSa2. Richard H. Lyon and his contributions to noise control.** George C. Maling (INCE Foundation, 102 Acorns Way, Brunswick, ME 04011, maling@alum.mit.edu)

Richard H. Lyon and I had similar interest in modal structure, his in rooms and in structures, and mine in reverberation rooms designed for the determination of sound power. He always had a keen interest in the sound radiated by complex structures. This led to the idea that the sound quality of a product could be influenced by careful adjustment of the many sound sources in a complex machine. He made many contributions to sound quality over the years. He had many other technical interests too numerous to list here, but he was the "father" of statistical energy analysis for which he was greatly admired. Later in life, we were both elected to the National Academy of Engineering (NAE). One tradition of the NAE is to prepare memorial tributes to deceased members for publication in an annual volume. I was pleased to collaborate with a colleague, David Bowen of Acentech Incorporated, on the tribute which detailed many of his contributions to acoustics and noise control.

1:50

**3pNSa3. Progress in product sound quality.** Patricia Davies (Ray W. Herrick Labs., Purdue Univ., 177 S Russell St., West Lafayette, IN 47907, daviesp@purdue.edu)

R.H. Lyon was a pioneer in several areas of acoustics, vibrations and noise control. His development of sound quality as an integral part of noise control engineering and product sound design is just one of many examples of his significant contributions to areas of importance in noise control engineering. In this presentation an overview of the Product Sound Quality workshops he organized in 1997 and 2002 will be given together with a summary of how the field has developed over the last 20 years, due, in no small part, to his leadership. The presentation will conclude with some personal memories of working with him.

WEDNESDAY AFTERNOON, 9 DECEMBER 2020 — 2:50 P.M. TO 4:00 P.M.

## Session 3pNSb

## Noise, Psychological and Physiological Acoustics, and Structural Acoustics and Vibration: Perception of Vehicle Noise II

Patricia Davies, Cochair
*Ray W. Herrick Laboratories, Purdue University, 177 S Russell Street, West Lafayette, IN 47907-2099*

Daniel Carr, Cochair
*Purdue University, 177 S. Russell Street, West Lafayette, IN 47907*

**Chair's Introduction—2:50**

### *Invited Papers*

2:55

**3pNSb1. An investigation of automatic flagging methods for aspiration noise in automobiles.** Daniel Carr (Purdue Univ., 177 S. Russell St., West Lafayette, IN 47907, djcarr@purdue.edu)

If there are small gaps in the seal between a closed automobile door or window and the rest of the car, annoying noises may arise due to airflow through the gaps, or aspiration. While some aspiration noises contain clearly identifiable tones or narrowband components, many aspiration noises do not. As a result, it is often difficult to identify aspiration noise, whether by listening to the recorded sound, or by examining the measured spectrum. Aspiration noise is classified as an error-state, which manufacturers want to remedy before selling the car, so an automatic tool for flagging sounds containing aspiration noise would be helpful. A method for simulating aspiration noise was developed, in which stationary and intermittent components are added to a "base sound" having a spectrum representative of a certain automobile. These simulated sounds were analyzed using trends of single-value metrics and statistics such as Sharpness and estimated kurtosis, and spectral measures such as the power spectral density of time-varying Loudness. The trends observed in these quantities, and their effectiveness as robust aspiration-noise flags, are reported. Future work may include developing a holistic flagging tool for all wind noise error-states, including whistles, cavity noise, and separated flow turbulence noise.

### 3:15

**3pNSb2. Assessments of vehicle interior noises—Considerations about the quality of listening test data.** Andre Fiebig (Eng. Acoust., TU Berlin, Einsteinufer 25, Berlin 10587, Germany, andre.fiebig@tu-berlin.de)

As there is a need for quantifying the perceived quality of automotive sounds in order to define target sound characteristics and set-up development targets, diverse jury test designs are frequently applied to collect data. The jury tests range from structured listening tests performed under controlled laboratory conditions to qualitative methods collecting free verbalization data during test drives. Though, jury tests are usually only used as an intermediate step to derive sound quality metrics based on psychoacoustic quantities allowing the quantitative prediction of subjective responses to product sounds. However, the quality of jury test data and the way of processing the data drive the validity of the developed sound quality metric. Therefore, a detailed quality check of listening test data is of particular importance but often neglected in practice. The quality check involves the investigation of intra- and inter-rater reliability issues, the detection of outliers and clusters, the use of adjusted error measures, the (cross) validation of sound quality metrics for estimating the generalizability of results. The different aspects in investigating listening test data in the context of vehicle noises and their suitability for sound quality metric development will be discussed on the basis of selected examples.

### 3:35

**3pNSb3. The role of dissonance in the sound quality perception for electric vehicles.** Arne Oetjen (Acoust. Group, Carl von Ossietzky Univ. Oldenburg, Carl-von-Ossietzky-Straße 9-11, Oldenburg D-26129, Germany, arne.oetjen@uni-oldenburg.de), Anna Rieger, Hans-Peter Rabl (Labor Verbrennungsmotoren und Abgasnachbehandlung, Ostbayerische Technische Hochschule Regensburg, Regensburg, Germany), and Steven van de Par (Acoust. Group, Carl von Ossietzky Univ. Oldenburg, Oldenburg, Germany)

Tonal components can be regarded as one of the most relevant components for interior sound quality of electric vehicles, especially at low speeds. Sources for these tones are for example the vehicles' drivetrains but also other technical components. Depending on the driving conditions, due to the interaction of various dominant sources, audible tone complexes can occur. Similar to musical intervals, the different frequency ratios within these complexes can result in consonant or dissonant sensations. The strength of this sensation does not only depend on the frequency relation of the tones but also on their relative levels. In this study, different musical intervals were subjectively rated in terms of their specific consonant or dissonant character. Using an adaptive procedure, sound pairs, each consisting of a two-tone signal, where adjusted to equal dissonance by varying the level of the second tone of one of the two-tone signals. Using the same paradigm, also pairs of equal vehicle sound quality were obtained. With these results, it is possible to quantify dissonance, and the influence of dissonance on sound quality expressed in terms of level differences of two-tone signals. These results can help to make the concept of dissonance more useful for sound-engineering applications.

WEDNESDAY AFTERNOON, 9 DECEMBER 2020 — 1:05 P.M. TO 2:35 P.M.

# Session 3pPAa

## Physical Acoustics: General Topics: Acoustic Characterization of Materials II

Colby Cushing, Chair
*Applied Research Laboratories, the University of Texas at Austin,*

**Chair's Introduction—1:05**

## *Contributed Papers*

### 1:10

**3pPAa1. Resonant ultrasound spectroscopy of cylindrically shaped sample of made of Ni-containing metallic glass alloy.** Oleksiy Svitelskiy (Phys., Gordon College, 255 Grapevine Rd., Wenham, MA 01938, oleksiy.svitelskiy@gordon.edu), Alexey Suslov (National High Magnetic Field Lab., Tallahassee, FL), Danielle Duggins, David Lee, and Kristen Siaw (Phys., Gordon College, Wenham, MA)

Among different methods for determining elastic constants of a material, a special place belongs to the resonant ultrasound spectroscopy (RUS), as to a fast and non-destructive technique. We use a home-made RUS spectrometer to investigate the elastic properties of $Ni_{71.5}Cr_{5.6}Nb_{3.4}P_{16.5}B_3$, a bulk metallic glass alloy (Ni-BMG). The sample was made by a counter-gravity suction casting technique and has shape of a cylinder. Modeling the measured spectrum of acoustic resonances for the sample in its cylindrical geometry was done with finite element analysis (FEA). FEA gives a longitudinal modulus ($C_{11}$) of 300 GPa, and a Young's modulus ($C_{44}$) of 50 GPa +/−5%. The determined values we verify by ultrasound pulse-echo technique, and compare them with the literature data for other Ni-bearing bulk metallic glass alloys. [This work was partially supported by NSF DMR award

number 1709282. National High Magnetic Field Laboratory is supported by NSF/DMR-1644779 and the State of Florida.]

1:30

**3pPAa2. The elastic constants measurements in a TI-6AL-4V eli alloy by ultrasound.** Hector Carreon (Universidad Michoacana, Santiago Tapia 403, Col. Centro UMS300101KE8, Morelia, Mich 58000, Mexico, hcarreon@umich.mx)

In this paper, we report the calculation of the elastic constants such Poisson's ratio, elastic modulus and shear modulus deduced from the ultrasonic propagating equations with two types of ultrasonic vibrations named longitudinal and transverse waves. The ultrasonic propagating velocity is measured at the microstructural evolution in a Ti-6Al-4V medical alloy with two varying microstructures, bimodal and acicular respectively. The two different initial microstructures were treated thermally by aging at 515 °C, 545 °C, and 575 °C at different times from 1 min to 576 h to induce a precipitation process. Ultrasonic measurements of shear and longitudinal wave velocities, scanning electron microscopy (SEM) image processing and micro-hardness were performed, establishing a direct correlation with the measurements of the ultrasonic velocity and the elastic properties developed during the thermal treatment of the artificial aging. The results of the ultrasonic velocity show a very clear trend as the aging time progresses, which is affected by precipitation of $Ti_3Al$ particles inside the α phase. In this way, we can know, in a fast and efficient way, the elastic properties developed during the heat treatment of aging at long times, since the presence of these precipitates hardens the material microstructure affecting the final mechanical properties.

1:50

**3pPAa3. Von Karman spatial correlation functions for modeling ultrasonic scattering in metallic media.** Andrea P. Arguelles (Eng. Sci. and Mech., Penn State Univ., 212 Earth-Engr Sci. Bldg, University Park, PA 16802, aza821@psu.edu)

In recent years, ultrasonic scattering has been studied as a potential method for nondestructive microstructure characterization in polycrystalline media. In order to invert microstructural data from ultrasonic data, analytical models of wave propagation and material statistics are generally employed. The microstructure statistics are frequently represented by spatial correlation functions (SCFs), which describe how microscopic variables at random positions are correlated (e.g., elastic stiffness). In polycrystalline media with statistically isometric crystallites, SCFs are defined as the probability that two randomly chosen points lie within a single crystallite (or grain). This work evaluates the suitability of common forms of the SCF in capturing the statistics of realistic microstructures in the context of ultrasonic scattering measurements. In addition, we introduce the von Karman SCF, frequently used in seismic wave propagation studies, as a potential alternative for metallic components with increased microstructural complexity. In the macroscale, the material systems are assumed to be statistically isotropic, and the microstructural differences focus on the morphology of the grain structure.

2:10

**3pPAa4. From 2D to 3D: Exploring the validity of two-point correlation functions to study polycrystalline media.** Tanni Alam (Eng. Sci. and Mech., Penn State Univ., 212 Earth and Eng. Sci., University Park, PA 16802, tja5483@psu.edu) and Andrea P. Arguelles (Eng. Sci. and Mech., Penn State Univ., University Park, PA)

Acoustic attenuation in polycrystalline materials is generally formulated using statistical representations of the material in the form of two-point correlation functions. Common formulations assume decoupling of the spatial and tensorial components and invoke analytical forms to represent the spatial correlation function (W(r)). A widely accepted form of W(r) is an exponential expression that employs a single characteristic length to describe the microstructure. More recently, this approach has been extended to include grain size distributions and different grain morphologies. To date, experimental approaches have relied on using 2D micrographs of material sections to obtain the necessary material parameters. However, the accuracy of the 2D statistical parameters relative to the three-dimensional two-point statistics is yet to be validated. In this presentation, we propose an analytical approach to transform 2D image data to 3D volumetric data by extending Wicksell's corpuscle problem to the case of equiaxed grains in polycrystalline materials. We use digital microstructures to evaluate the validity of the transformation for a broad scope of grain size distribution parameters and evaluate the accuracy of the analytical forms of the spatial correlation function. Finally, we study the impact of these approximations on acoustic attenuation.

## Session 3pPAb

# Physical Acoustics: General Topics: Acoustic Characterization of Materials III

Colby Cushing, Chair
*Applied Research Laboratories, the University of Texas at Austin,*

**Chair's Introduction—2:50**

## *Contributed Papers*

**2:55**

**3pPAb1. Ultrasonic properties of Inconel 718 fabricated via laser-directed energy deposition.** Guillermo E. Huanes-Alvan (Mech. Eng., Michigan State Univ., 428 S Shaw Ln., Rm. 2222/Rm. B310, East Lansing, MI 48824, huanesal@egr.msu.edu), Beytullah Aydogan, Himanshu Sahasrabudhe (Mech. Eng., Michigan State Univ., East Lansing, MI), and Sunil Kishore Chakrapani (Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI)

Laser-directed Energy Deposition (laser-DED) is an additive manufacturing process where a 3D structure is fabricated layer-by-layer with powder that is directed and melted using a high power laser. The processing parameters of laser-DED have a strong influence on the microstructure of the samples that are fabricated. Due to a very high cooling rate, laser-DED produces some unique microstructures like columnar, dendritic grains, and macro-micro grains. The objective of this study is to understand how these microstructures will influence the ultrasonic properties of Inconel 718 (IN718). This work focuses on frequency-dependent parameters such as ultrasonic phase velocity, attenuation coefficient, backscatter and absorption, since they can be related to microstructural features spread over multiple wavelengths: grain size, grain orientation, phases, etc.

**3:15**

**3pPAb2. Simultaneous high-speed ultrasound and synchrotron x-ray imaging for monitoring melt pool dynamics in metallic additive manufacturing.** Jared Gillespie (Eng. Sci. and Mech., The Penn State Univ., 212 Earth and Eng. Sci. Bldg, University Park, PA 16802, jvg5970@psu.edu), Wei Yi Yeoh, Bo Lan (Mech. Eng., Imperial College London, London, United Kingdom), Tao Sun (Mater. Sci. & Eng., Univ. of Virginia, Charlottesville, VA), and Christopher M. Kube (Eng. Sci. and Mech., The Penn State Univ., University Park, PA)

Fundamental to metallic additive manufacturing (AM) is the laser-powder-substrate interaction that leads to localized melting of metallic powder to a previously solidified substrate. The formation of the melt pool and the subsequent solidification dictate the resultant microstructures and properties of the processed materials, which then influence both part quality and performance. Open ended questions involving the optimization of process parameters for defect free and high performing parts remain at the heart of AM science. Further understanding of melt pool and solidification behavior is required to help answer these questions. In this presentation, 25 MHz ultrasonic shear waves are used to probe dynamic melt pool behavior in Al6061 by observing the scattering from the solid/liquid and liquid/gas boundaries. High-speed synchrotron x-ray imaging was employed simultaneously with the ultrasonic measurement for validation. Melt pool features observed in the x-ray images were then associated with features in the synced ultrasound measurement. Specifically, the scattered response tracked with the depth of the melt pool. Finite element models depicting wave-scattering were also used to predict wave scattering and help interpret the ultrasonic response. These preliminary results provide support for ultrasonic scattering as a promising method to evaluate melt pool behavior in metallic additive manufacturing processes.

**3:35**

**3pPAb3. Elastic properties of additively manufactured alloys.** Gabriela Petculescu (Univ. of Louisiana at Lafayette, PO Box 44210, Lafayette, LA 70504, gp@louisiana.edu), Damilola Dada, Naresh Deoli, Jonathan Raush (Univ. of Louisiana at Lafayette, Lafayette, LA), and Shengmin Guo (Louisiana State Univ., Baton Rouge, LA)

With the advent of metal Additive Manufacturing (AM), or 3D printing, the focus on evaluating and fine-tuning the fabrication process has become crucial. High-resolution volumetric mapping of elastic properties throughout an AM built sample can be done, "biopsy" style, with the highly adaptable Resonant Ultrasound Spectroscopy (RUS) technique. We report results of elastic constants measurements on AM-fabricated Ti64 and CL 80CU bronze (i.e., $Cu_{0.9}Sn_{0.1}$) alloy with RUS. Despite an easily observable pore distribution, surprisingly sharp spectra and good RUS fits were obtained for both. The average $E$ and $G$ moduli obtained for AM Ti64 at room temperature are $\approx$104 and $\approx$39.4 GPa, respectively. The values are only a few percent lower than those of the traditionally manufactured alloy. Moreover, stress-strain measurements on dog-bone samples of Ti64 from the sample batch lead to a value of 101 GPa for E. Heat-treatment of as-manufactured Ti64 samples lead to a slight increase in both moduli. For the investigated bronze, an $\approx$10% variation of elastic properties within a cross section of a printed sample was observed. The average $E$ and $G$ moduli obtained for AM CL 80CU bronze at room temperature are $\approx$111 and $\approx$40.9 GPa, respectively.

*Invited Paper*

3:55

**3pPAb4. Experimental determination of the elasticity of porous ceramics under the gas saturation and temperature treatment.**
Ashoka Karunarathne (Phys. and Astronomy & NCPA, Univ. of MS, 145 Hill Dr., University, MS 38677, atholang@go.olemiss.edu) and Joseph Gladden (NCPA / Office of Res., Univ. of MS, University, MS)

After the formulation of the theory of poroelasticity, which describes the elasticity of fluid saturated porous media, many experimental studies have been reported on the determination of the elasticity of porous materials under the various physical and structural conditions. However, there are limited studies on how the elasticity behave under the variation of the physical conditions of the saturated fluid. This study presents an attempt to characterize the influence of the pore-saturated gas media and their physical properties on the elasticity of porous ceramic materials. Resonant ultrasound spectroscopic measurements were performed on test specimens of alumina with ∼40% porosity, zirconia with ∼48% porosity, and sintered fully dense zirconia to determine the hydrostatic pressure-dependent macroscopic elasticity. Here, we report the variation of elasticity of porous and full dense samples over approximately five orders of magnitude (800–0.02 psi) in absolute pressure. A material softening with increased pressure was observed from the porous specimens and the rate of softening is quantified with different gas saturation by helium, nitrogen and argon. The time evolution of mechanical equilibrium of the porous materials at low pressure (0.02 psi) and moderately high-temperature (150 °C) conditions will also be discussed.

WEDNESDAY AFTERNOON, 9 DECEMBER 2020 — 1:05 P.M. TO 2:35 P.M.

## Session 3pPP

## Psychological and Physiological Acoustics: Localization and Unmasking

Erol J. Ozmeral, Chair
*Communication Sciences and Disorders, University of South Florida, 4202 E. Fowler Avenue, Tampa, FL 33620*

**Chair's Introduction—1:05**



*Contributed Papers*

1:10

**3pPP1. New measurements of the frequency weighting for localising speech in noise.**
Michael A. Akeroyd (School of Medicine, Univ. of Nottingham, Hearing Sci. Bldg., University Park, Nottingham NG7 2RD, United Kingdom, michael.akeroyd@nottingham.ac.uk) and Jennifer Firth (School of Medicine, Univ. of Nottingham, Nottingham, United Kingdom)

It is generally noted that for the purposes of locating sounds, low-frequency ITDs are dominant. Much of the data for this come from HRTF experiments with noise signals. Here, we apply the "revcorr" technique to determine which frequencies are dominant for localising freefield speech. We filtered brief signals (either the start of, end of, or complete single words) into 10 channels using a gammatone filterbank (3-ERB spacing from 153 to 6780 Hz). We randomly divided the filtered signals into two groups (e.g., channels 1-2-6-7 + 3-4-5-8-9-10; or 4-6-8-9-10 + 1-2-3-5-7, etc.) then played, simultaneously, one group from a loudspeaker at −30 deg and the other from +30 deg. A spectrally shaped diffuse masking noise was mixed at a per-channel signal-to-noise ratio of +20 dB. The task was to decide where the perceived speech signal was. By running thousands of trials per listener, randomly choosing the +30/−30 grouping of channels each time, we were able to determine which frequencies were weighted most in determining the location of the speech signal. Nine listeners have completed the task so far. We found that, contrary to most previous results, high frequencies are weighted more than low frequencies.

1:30

**3pPP2. The role of lateralization in release from informational masking.**
Richard Freyman (Commun. Disord., Univ. of Massachusetts, 358 N. Pleasant St., Amherst, MA 01106, rlf@comdis.umass.edu), Kaitlyn Denson, Amber Fields (Commun. Disord., Univ. of Massachusetts, Amherst, MA), Emily Buss (Otolaryngology/Head and Neck Surgery, The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), and Patrick Zurek (Sensimetrics Corp., Woburn, MA)

This study tested whether target and masker must create different spatial percepts, in this case intracranial percepts created under headphones, to achieve spatial release from informational masking. Signals were sequences of 40-ms broadband noise bursts with four target bursts interspersed within a longer train of masker bursts. From a baseline sequence in which target bursts were 400 ms apart, either the second or third burst was advanced by 80 ms to create two different temporal patterns that subjects discriminated.

Target bursts were intermixed among masker bursts such that pattern identification was at chance when all bursts were diotic, but near perfect when target and masker bursts were delivered with moderate interaural differences of opposite sign. Performance in these unprocessed conditions was compared to that obtained when stimuli were processed by an algorithm that swapped interaural differences in alternating frequency bands [S. Sheffield *et al.*, *J. Acoust. Soc. Am.* **145**, 1129–1142 (2019)], thereby degrading lateralization but preserving the magnitudes of interaural differences. Early results show substantially reduced masking release in degraded lateralization conditions, despite their preservation of interaural differences, suggesting a prominent role for perceived target-masker separation in spatial release from informational masking. [Work supported by NIDCD R01 01625.]

**1:50**

**3pPP3. Assessment of dynamic spatial release from masking via listener head rotation.** Ewan A. Macpherson (National Ctr. for Audiol., Western Univ., 1201 Western Rd., EC 2262, London, ON N6G 1H1, Canada, ewan.macpherson@nca.uwo.ca)

The dynamic interaural time-difference (ITD) created by listener head rotation is a potent cue for front/rear sound source localization. In principle this cue could also assist in the segregation of simultaneously presented front and rear sources, particularly when robust high-frequency spectral cues are absent. If so, head rotation might provide dynamic spatial release from masking. We assessed this in a spatial auditory attention task in which multiple different equal-intensity sequences of four spoken digits, low-pass filtered at 1500 Hz, were presented simultaneously—the target sequence from 0 or 180 deg azimuth and distractors from lateral angles of $\pm$22.5 and/or $\pm$45 deg relative to the target, but in the opposite hemisphere. On each trial, listeners either fixated towards 0 azimuth or oscillated their heads at $\sim$0.5 Hz with an amplitude of $\sim\pm$40 deg. Listeners reported the target sequence heard. In a majority of listeners, there was no benefit of head motion, and therefore no evidence of dynamic spatial release from masking for these stimuli. These results are consistent with those of Culling [*J. Exp. Psych.* **26**, 1760–1769 (2000)], who found that in tone complexes with components smoothly changing in ITD, opposite movement direction for one component was not an effective segregation cue.

**2:10**

**3pPP4. Walking while listening and remembering: Dual- and multi-task costs.** Karen Helfer (Commun. Disord., Univ. of Massachusetts, 358 N. Pleasant St., Amherst, MA 01003, helfer@umass.edu), Richard Freyman (Commun. Disord., Univ. of Massachusetts, Amherst, MA), Richard van Emmerik, Jacob Banks (Kinesiology, Univ. of Massachusetts, Amherst, MA), Michael Clauss, Lincoln Dunn, and Silvana Tellerico (Commun. Disord., Univ. of Massachusetts, Amherst, MA)

In day-to-day life people often need to perform more than one task at a time. Dual- and multi-task costs may occur in these situations. In the present study we examined the realistic situation in which people must communicate in an acoustically-challenging environment while walking. Participants (younger adults and middle-aged adults) performed a task of listening and responding to speech presented in competing speech. In this task, participants repeated back sentences presented from a front loudspeaker in the presence of two-talker same-sex speech maskers presented from loudspeakers to the right and left. In some conditions they were asked to also complete a memory task in which they were shown a sequence of four numbers and letters (e.g., "C9L3") before the trial began and had to report that sequence at the end of the trial. These tasks were completed while either standing still or while walking on a treadmill. Gait parameters were measured using sensors attached to participants' wrists, legs, waist, chest, and forehead. This presentation will detail results of preliminary analyses designed to examine costs involved in performing these tasks simultaneously compared to individually in the two age groups. [Work supported by NIDCD 012057.].

WEDNESDAY AFTERNOON, 9 DECEMBER 2020 — 12:00 NOON TO 12:45 P.M.

## Session 3pSCa

## Speech Communication, Psychological and Physiological Acoustics, Noise, and Architectural Acoustics: Listening in Challenging Circumstances III (Poster Session)

Authors will be at their posters from 12:00 noon to 12:45 P.M.

### *Contributed Papers*

**3pSCa1. Impact of phase distortion and phase-insensitive speech enhancement on speech quality perceived by hearing-impaired listeners.** Zhuohuang Zhang (Dept. of Speech, Lang. and Hearing Sci., Indiana Univ. Bloomington, 200 South Jordan Ave., Bloomington, IN 47405, zhuozhan@iu.edu), Donald S. Williamson (Dept. of Comput. Sci., Indiana Univ. Bloomington, Bloomington, IN), and Yi Shen (Dept. of Speech, Lang. and Hearing Sci., Indiana Univ. Bloomington, Bloomington, IN)

Phase is important for speech since it contributes to the quality and intelligibility during speech perception. Many speech enhancement algorithms lack the ability to predict phase for speech reconstruction and apply the noisy phase instead. In this study, we investigated the influence of phase distortion on the speech-quality ratings of both normal-hearing (NH) and hearing-impaired (HI) listeners by applying different degrees of random phase on speech. In one set of conditions, the speech was embedded in babble noise at 4 different signal-to-noise ratios (SNRs) from -5 to 10 dB, while in another set of conditions, the SNR was fixed at 10 dB and the speech and noise mixture was presented in simulated rooms with reverberation times ranged from 100 to 1000 ms. The speech level was kept at 65 dB SPL for NH listeners; while amplification was applied to ensure the audibility for HI listeners. Ideal ratio mask (IRM) was used for speech enhancement. Speech-quality ratings were collected following the MUSHRA procedure and compared to two objective evaluation metrics (i.e., PESQ and HASQI). Results suggest that phase distortion has negative impact on speech quality for both NH and HI listeners and this effect becomes dominant when speech enhancement was applied.

**3pSCa2. The role of attention in listening-in-noise.** Rochelle S. Newman (Dept of Hearing & Speech Sci., Univ. of Maryland, 0100 Lefrak Hall, College Park, MD 20742, rnewman1@umd.edu), Monita Chatterjee (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE), Karen Mulak (Hearing & Speech Sci., Univ. of Maryland, College Park, MD), Janet Frick (Psych., Univ. of Georgia, Athens, GA), and Emily Shroads (Hearing & Speech Sci., Univ. of Maryland, College Park, MD)

Listeners of all ages are faced with having to understand speech in the presence of noise. One factor that might influence listeners' success at this task is their ability to attend selectively. We examine both adults' and toddlers' understanding of speech in the presence of nonlinguistic background noise – noise that is either easier or harder to attend to selectively. We also test the same individuals on a nonlinguistic selective attention task. Adult listeners easily take advantage of acoustic cues such as spatial location and frequency-range differences to help them attend selectively to, and comprehend, the target speech. Young toddlers (under 2 years of age) fail to use these cues. In addition, among adults, success at listening in noise is correlated with performance in a visual selective attention task, suggesting that domain-general attention skills are relevant for adults' listening in noise. This is less clearly the case for young children, however; indeed, children who were more easily distracted by extraneous sounds showed better speech understanding in noise. These findings suggest that the relationship between performance in challenging listening conditions and other cognitive skills undergoes critical changes across development.

**3pSCa3. Exploring the effects of phonetic overlap and background noise on incremental processing in children.** Y. Sophia Liu (Northwestern Univ., 633 Clark St., Evanston, IL 60208, yishuliu2021@u.northwestern.edu), Katherine Simeon, and Tina Grieco-Calub (Northwestern Univ., Evanston, IL)

In everyday conversation, individuals actively process speech in order to comprehend and respond in real-time. As a word unfolds, listeners activate possible lexical candidates and actively determine the target word as soon as possible. This process requires knowledge of one's native spoken language and the ability to recognize individual phonemes. Phonetic overlap between the target and candidate words can influence the speed of lexical access. For example, phonological onset (e.g., *candy*), but not rhyming words (e.g., *sandal*), can yield lexical competition with target words (e.g., *candle*). When the auditory input is spectrally degraded, however, the effect is different: rhyme words have greater lexical competition with target words. The purpose of the present study is to test the extent to which rhyme words compete with target words in the present of background noise. In a visual world paradigm, the present study examines the time course of spoken word recognition when target words (e.g., *candle*) are contrasted with full (e.g., *sandal*) and offset (e.g., *poodle*) rhymes in 5-to-7-year-olds. Target words are presented in both quiet and in steady-state speech-shaped noise at 0 dB SNR. Preliminary data suggest that rhyming overlap does not impact processing in quiet but may show differences in noise.

**3pSCa4. Talker depressive symptoms affect intelligibility of noise-adapted speech.** Hoyoung Yi (Speech, Lang., and Hearing Sci., Texas Tech Univ. Health Sci. Ctr., 3601 4th St. Stop 6073, Lubbock, TX 79430-6073, hoyoung.yi@ttuhsc.edu) and Rajka Smiljanic (Univ. of Texas at Austin, Austin, TX)

A critical aspect of communicative competence is talker's ability to adapt the clarity of their speech in response to the challenges of a particular communicative situation. Previous work showed that talkers with elevated depressive symptoms were less successful in producing listener-oriented intelligibility-enhancing clear speech. The present study investigated intelligibility of speech produced in response to environmental noise (noise-adapted speech, NAS) in talkers with high depressive (HD) and low depressive (LD) symptoms. Sentence intelligibility was examined in the presence of speech-shaped noise (SSN) and 1-talker (1T) competing speech. Results revealed that NAS increased intelligibility for both talker groups and in both maskers. NAS intelligibility benefit was smaller for HD talkers than LD talkers in the SSN condition. Acoustic analyses showed that NAS was characterized by a decreased speaking rate, increased F0 mean and range, and increased energy in the 1–3 kHz range. Talkers with HD symptoms however produced these modifications significantly less compared to talkers with LD symptoms. Results provide further evidence that elevated depressive symptoms impact speaking style adaptations in response to various communicative barriers leading to lower intelligibility. The results have the potential to aid in clinical decision making for individuals with depression.

**3pSCa5. Rapid adaptation to non-native speech: Effects of aging, hearing loss, and stimulus variability.** Rebecca E. Bieber (Univ. of Maryland, College Park, 0119 Lefrak Hall, 7251 Preinkert Dr., College Park, MD 20740, rbieber@umd.edu) and Sandra Gordon-Salant (Univ. of Maryland, College Park, College Park, MD)

When communicating in challenging situations, most younger listeners with normal hearing are able to quickly adapt to the speech signal, improving their recognition with exposure. One potentially challenging situation is communication with non-native speakers whose second-language production (i.e., accented speech) is altered by the spectral and temporal characteristics of the L1. Older listeners, particularly those with age-related hearing loss, report considerable difficulty in recognition of non-native speech. A growing body of literature suggests that older adults can also perform rapid adaptation to non-native speech, but the conditions that might hinder or facilitate adaptation for these listeners remain unclear. In the present study, rapid adaptation was evaluated for three groups of listeners (younger with normal hearing, older with normal hearing, and older with hearing impairment) in conditions with increasing levels of stimulus variability. Generalization to unfamiliar talkers with familiar and unfamiliar accents was also assessed. Preliminary results indicate differing patterns of adaptation and magnitude of adaptation across conditions and listener groups. Generalization to an unfamiliar talker with a familiar accent appears greater following exposure to multiple talkers than following exposure to a single talker. Generalization to an unfamiliar talker with an unfamiliar accent appears limited.

**3pSCa6. Effect of envelope signal-to-noise ratio on the intelligibility of speech in speech-spectrum shaped noise.** Rahim Soleymanpour (Biomedical Eng., Univ. of Connecticut, 263 Farmington Ave., Farmington, CT 06030, rahim.soleymanpour@uconn.edu), Anthony J. Brammer, and Insoo Kim (Dept. of Medicine, Univ. of Connecticut, Farmington, CT)

In attempts to improve speech intelligibility, the envelope of speech in noise is commonly processed in the time domain and used to amplitude modulate the corrupted speech. In this study the envelope was constructed from the noisy speech, but separately and independently of the SNR of the noisy speech. Envelopes corresponding to 0, −3, −6, and −9 dB SNR were applied to the noisy speech when the latter had an SNR of either −9, −6, −3, or 0 dB, to evaluate the benefit to intelligibility of improving the SNR in the modulation domain. Signals from 200 Hz to 6 kHz were processed by MATLAB into sixteen contiguous subbands with bandwidth approximately 1.5×ERB of an auditory filter. The subband envelopes were formed from the absolute value of the signals and low-pass filtered. Eleven subjects aged $29 \pm 8$ years (mean and range) with normal hearing underwent the Modified Rhyme Test to assess speech intelligibility. The stimuli were presented diotically over earphones with the subject seated in an audiometric room. Statistically significant increases in mean word score of up to 35% could be obtained by improving the envelope SNR, suggesting this processing may benefit speech intelligibility. [Work supported by NIOSH.]



**3pSCa7. Subphonemic variation detection and correction in cochlear implant users.** Sarah Bakst (Commun. Sci. and Disord., Univ. of Wisconsin–Madison, 1500 Highland Ave., Madison, WI 53705, sbakst@wisc.edu), Caroline A. Niziolek, and Ruth Y. Litovsky (Commun. Sci. and Disord., Univ. of Wisconsin–Madison, Madison, WI)

The ability to detect and correct errors in one's own productions is crucial for intelligible speech. Individuals with normal hearing detect and correct even within-phoneme deviations while talking. Cochlear implant (CI) users receive spectrally-degraded auditory information, potentially filtering out subphonemic information and prohibiting error detection. The experiment here tested the threshold at which CI users could detect variation in the first and second formants (F1/F2), spectral peaks which are important

for distinguishing vowel height and backness. We hypothesized that CI users may not be able to detect subphonemic differences in their speech, especially if those differences do not cross wide electrode boundaries in their frequency allocation tables. In an adaptive discrimination task, speakers distinguished versions of their own speech where F1 or F2 had been altered. Contrary to the hypothesis, some speakers were sensitive to acoustic differences that occurred within a single electrode band, suggesting that speakers may be able to use subphonemic information to constrain acoustic variability while talking. We will also present pilot data from a real-time altered auditory feedback experiment. Preliminary evidence showing opposition to the formant shift suggests that CI users may have access to and use acoustic detail to correct errors while talking.

**3pSCa8. Diverse environments and their impact on accentedness judgments.** Ethan Kutlu (Linguist., Univ. of Florida, 3705 Southwest 27th St., Unit 623, Gainesville, FL 32608, ethankutlu@outlook.com), Mehrgol Tiv (Psych., McGill Univ., Montreal, QC, Canada), Stefanie Wulff (Linguist., Univ. of Florida, Gainesville, FL), and Debra Titone (Psych., McGill Univ., Montreal, QC, Canada)

Research shows that listeners' perceived accentedness can be mediated by visual input (Babel & Russell, 2015; McGowan, 2015; Zheng & Samuel, 2017), and can change depending on their exposure to varied speech (Baese-Berk *et al*., 2013). Here, we tested the impact that visual input and linguistic diversity has on listeners' perceived accentedness judgments. Two experiments were conducted: one in Gainesville (USA) and one in Montreal (Canada). While these two locations were selected for their bilingual populations, Montreal has a more diverse linguistic landscape (Gullifer & Titone, 2019). Participants completed an accentedness judgment task where they were shown either a White or a South-Asian face while listening to sentences in American, British, and Indian English. They also completed a language background questionnaire, a social network questionnaire, and executive control tasks. In an ongoing study, results show that for Gainesville, participants' perceived accentedness of all three varieties increased when the visual input changed from a White face to a South-Asian face ($F(2, 66) = 33.3$, $p < 0.001$). The same effect was not observed for listeners in Montreal ($F(2,23) = 0.664$, $p = 0.524$). These preliminary findings suggest that exposure to both different accents and racial/ethnic categories on a regular basis could impact perceived accentedness judgements.

**3pSCa9. Speech intelligibility in primary school children: A systematic review.** Silvia Murgia (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 South Sixth St., Champaign, IL 61820, smurgia2@illinois.edu), Jossemia Webster (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL), Lady Catherine Cantor Cutiva (Universidad Nacional de Colombia, Bogota, Colombia), and Pasquale Bottalico (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Good quality of voice signal and adequate classroom acoustics are important factors to guarantee an optimal teaching-learning process. This systematic review aims to characterize the relationship between intelligibility of the speech and room acoustics in primary schools based on the available evidence. Eligible studies were identified using two computerized databases: PubMed and Scopus. In total, 21 publications met our inclusion criteria: (1) subjects must have been from primary schools; (2) acoustic characterization of the classroom must been provided; (3) intelligibility tests must have been performed; and (4) articles written in English. After identifying the parameters and tests used to define the intelligibility of the speech, the subjective scores were analyzed in relationship with voice related parameters found in the articles (SNR and STI). Our results highlight the negative effect on intelligibility associated with the bad transmission of the speech and a poor acoustic of the classroom caused by long reverberation times and high background noise. Future research needs to include stronger study designs than "one-time survey" and analyze the results even in cases of acoustic interventions.

WEDNESDAY AFTERNOON, 9 DECEMBER 2020 1:05 P.M. TO 1:50 P.M.

## Session 3pSCb

## Speech Communication, Architectural Acoustics, Psychological and Physiological Acoustics, and Noise: Listening in Challenging Circumstances IV (Poster Session)

Authors will be at their posters from 1:05 p.m. to 1:50 p.m.

### *Contributed Papers*

**3pSCb1. Investigation of a temporal modulation based method on the intelligibility of speech in speech-spectrum shaped noise.** Rahim Soleymanpour (Biomedical Eng., Univ. of Connecticut, 263 Farmington Ave., Farmington, CT 06030, rahim.soleymanpour@uconn.edu), Anthony J. Brammer, and Insoo Kim (Dept. of Medicine, Univ. of Connecticut, Farmington, CT)

Temporal modulation is frequently considered for improving the intelligibility of noisy speech owing to the importance of the envelope in conveying speech information. In this study, we evaluated the performance of temporal modulation on word scores obtained with speech-in-noise. We firstly divided the noisy speech into 16 contiguous subbands from 200 Hz to 6 kHz with bandwidth approximately 1.5xERB of an auditory filter. The temporal envelope was constructed for each subband from the absolute value. It was then low-pass filtered at 16 Hz and used as the instantaneous gain for the subband's noisy speech. We established a psychophysical test (Modified Rhyme Test) with different speech-in-noise SNR values of 0, −3, −6, and −9 dB. Eleven native speakers (age $29 \pm 8$) with normal hearing were recruited for the study. The signals were processed by MATLAB and presented over earphones to participants seated in an audiometric room. Comparing the processed and unprocessed noisy speech, the mean differences in word scores for SNRs of 0, −3, −6, and −9 dB were −0.4%, 1.5%, −10.5%, and 1.4%, respectively. Using ANOVA, we concluded that the temporal modulation-based algorithm does not produce a statistically significant improvement in speech intelligibility. [Work supported by NIOSH.]

**3pSCb2. Accent familiarity and adverse listening conditions.** Christina M. Sen (School of Speech, Lang., and Hearing Sci., San Diego State Univ., SLHS Bldg. Rm. 221, 5500 Campanile Dr., San Diego, CA 92182-1518, csen7960@sdsu.edu) and Molly Babel (Linguist., Univ. of BC, Vancouver, BC, Canada)

Our ability to understand spoken language depends on both the speaker and listening environment. Speech intelligibility is higher in more favorable listening environments. Accent familiarity also plays a major role, with familiar accents being more intelligible than unfamiliar accents. Previous research [S. Gittleman and K. J. Van Engen, "Effects of noise and talker intelligibility on judgments of accentedness," *J. Acoust. Soc. Am.* **143**(5), 3138–3145 (2018)] found a significant relationship between the level of noise and perceived "foreignness" of an accent for Mandarin-accented English and locally accented English speakers. Their results show that when participants listened to Mandarin-accented speakers they rated them as less accented in lower signal-to-noise ratios (SNR). However, when participants listened to local English speakers, they rated the speakers as more accented in lower SNRs. The current study replicates these procedures using locally accented English ($n = 5$) and Hawaiian Pidgin English ($n = 5$). In a within-listener design, listeners complete an intelligibility task and an accentedness rating task. In the accent rating task, listeners are presented with sentences in four conditions: clear, +4 db SNR, 0 db SNR, and −4 db SNR. These results will contribute to our understanding of familiarity in speech processing.

**3pSCb3. Intelligibility of dysphonic speech in auralized classrooms.** Silvia Murgia (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 South Sixth St., Champaign, IL 61820, smurgia2@illinois.edu), Giuseppina Emma Puglisi (Dept. of Energy, Polytechnic of Turin, Torino, Italy), Tomas E. Sierra (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL), Arianna Astolfi (Dept. of Energy, Polytechnic of Turin, Torino, Italy), Keiko Ishikawa, and Pasquale Bottalico (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Voice disorders reduces speech intelligibility. This study evaluated the effect of noise, voice disorders and room acoustics on vowel intelligibility. Twenty-nine college students listened to 11 vowels in /h/-V-/d/ format. The speech was recorded by three adult females with dysphonia and three adult females with normal voice quality. The recordings were convolved with two oral-binaural impulse responses with 0.4 s and 3.1 s of reverberation time. The intelligibility and the listening easiness were significantly higher in quiet condition, when the speakers had normal voice quality and in low reverberated environments, while the response time of the listener was longer in noise condition.

**3pSCb4. Cochlear-implant simulation mimics increased age in temporal processing of syllables by young and middle-aged normal-hearing listeners.** Catherine L. Rogers (Commun. Sci. and Disord., Univ. of South Florida, PCD1017, Tampa, FL 33620, crogers2@usf.edu), Jenna Vallario, Morgan O'Malia (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL), and Gail S. Donaldson (Commun. Sci. and Disord., Appalachian State Univ., Boone, NC)

To better understand the effects of signal quality on phonetic processing, the present study examines the effects of signal degradation on processing of two-syllable sequences by young and middle-aged normal-hearing listeners. Temporal processing of American-English vowel sequences was compared across unprocessed and cochlear-implant (CI) simulation conditions for young listeners and for a CI simulation condition for the middle-aged listeners. Using the method of Fogerty, Humes and Kewley-Port [*J. Acoust. Soc. Am.* **127**, 2509–2520 (2010)], listeners heard 70-ms, resynthesized versions of four syllables ("pit, pet, put, pot") in a two-syllable temporal-order processing task. Task difficulty was adjusted by increasing or decreasing syllable-onset asynchrony (SOA), i.e., the duration between syllable onsets. SOA thresholds for accuracy of syllable-sequence identification was estimated using the method of constant stimuli on each of four 72-trial blocks. Results for the CI simulation conditions for both listener groups were comparable to those for the next older age-group in Fogerty *et al.* (2010), i.e., young adult listeners performed similarly to the middle-aged listeners Fogerty *et al.*, and middle-aged listeners performed similarly to the older listeners Fogerty *et al*. Results will be discussed with regard to implications for phonetic processing of speech in demanding listening conditions and practical implications for CI users.

**3pSCb5. Phonemic restoration in monolingual and bilingual listeners in a sentence context.** Erika L. Exton (Hearing and Speech Sci., Univ. of Maryland, College Park, 0100 LeFrak Hall, College Park, MD 20742, eexton@umd.edu) and Rochelle S. Newman (Hearing & Speech Sci., Univ. of Maryland, College Park, MD)

Phonemic restoration, the illusion in which listeners perceive a word as intact when a phoneme is replaced by non-speech noise, has been shown in both adults and children (Warren 1970; Newman 2004). Phonemic restoration appears to have a lexical basis (Samuel 1997); however, bilingual listeners may be limited in their ability to engage this top-down skill because their lexical knowledge is necessarily less robust in a given language than that of their monolingual peers. Bilingual phonemic restoration has rarely been shown, particularly in sentences, but exploring it allows us to better understand the top-down linguistic factors of the effect. In the present study, monolingual and bilingual English speakers listened to and transcribed 92 low-predictability sentences in four conditions: no noise, pink noise throughout, pink noise-filled gaps, and silent gaps. Here "bilingual" includes both crib bilinguals and English L2 speakers. Preliminary results suggest that monolingual listeners show the standard phonemic restoration effect such that transcription is more accurate for filled-gap than silent-gap sentences, but that bilingual listeners do not. For bilingual listeners only, English verbal fluency scores are positively correlated with transcription accuracy in all three degraded conditions (noise throughout, filled-gap, and silent gap), suggesting that language proficiency contributes to listeners' ability to comprehend speech in various types of noise.

**3pSCb6. Intelligible nonnative-accented speech incurs a processing cost: Converging evidence from a behavioral and a physiological paradigm.** Violet A. Brown (Psychol. & Brain Sci., Washington Univ. in St. Louis, 1 Brookings Dr., Apt. 5, St. Louis, MO 63130, violet.brown@wustl.edu), Drew J. McLaughlin (Psychol. & Brain Sci., Washington Univ. In St Louis, St. Louis, MO), Julia F. Strand (Psych., Carleton College, Northfield, MN), and Kristin J. Van Engen (Psychol. and Brain Sci., Washington Univ. in St. Louis, St. Louis, MO)

In noisy settings or when listening to an unfamiliar talker or accent, it may be difficult to understand spoken language. This difficulty can result in reductions in speech intelligibility, but may also increase the effort necessary to process the speech. In the current study we used a dual-task paradigm and pupillometry to assess the cognitive costs associated with processing fully intelligible accented speech, predicting that participants would expend greater effort when processing nonnative-accented speech, and that this cost would be attenuated over time. Consistent with our hypothesis, the behavioral and physiological paradigms provided converging evidence that listeners expend greater effort when processing nonnative- relative to native-accented speech. We also observed an overall reduction in listening effort over the course of the experiment and found some evidence for greater perceptual adaptation to nonnative-accented speech. These results suggest that even when speech is fully intelligible, resolving deviations between the acoustic input and stored lexical representations incurs a processing cost, and adaptation may attenuate this cost.

**3pSCb7. Insights into rapid adaptation to Spanish-accented English words in younger and older adults.** Rebecca E. Bieber (Univ. of Maryland, College Park, 0119 Lefrak Hall, 7251 Preinkert Dr., College Park, MD 20740, rbieber@umd.edu), Anna R. Tinnemore, and Sandra Gordon-Salant (Univ. of Maryland, College Park, College Park, MD)

Older adults often report difficulty understanding speech produced by non-native talkers. Although numerous studies have described auditory training protocols to improve speech recognition in noise, there are comparatively few studies evaluating auditory training for improving recognition of non-native speech. However, there is evidence that older adults can achieve short-term, rapid adaptation to non-native speech. In this study, a word-level training paradigm was employed, targeting improved recognition of Spanish-accented English. Younger and older adults with normal hearing or hearing impairment were trained on Spanish-accented monosyllabic word pairs containing four phonemic contrasts (initial s/z, initial v/f, final b/p, final d/t) produced in English by multiple male native Spanish speakers. Listeners completed pre-testing, training, and post-testing, over two sessions. While the training protocol failed to elicit any long-term learning for Spanish-accented speech, detailed examination of listeners' performance during the pre-testing revealed patterns of rapid adaptation that appear to vary by phonemic contrast and that may be dependent on the acoustic features of the stimulus. Statistical methods such as growth curve modeling and generalized additive mixed models are employed to describe the patterns of rapid adaptation, and how they vary between listener groups and different stimulus types.

**3pSCb8. Bilinguals' comprehension of foreign-accented speech.** Sita Carraturo (Psychol. & Brain Sci., Washington Univ. in Saint Louis, 1 Brookings Dr., Saint Louis, MO 63130, sita@wustl.edu) and Kristin J. Van Engen (Psychol. and Brain Sci., Washington Univ. in St. Louis, St. Louis, MO)

Foreign accents represent a common challenge to successful speech recognition, and while much has been uncovered about the factors contributing to the intelligibility of foreign-accented speech, most of it is based on data from monolingual listeners. There is good reason to believe, however, that bilinguals have a different experience when decoding foreign-accented speech: their two phonologies offer both greater flexibility in phonological-lexical mapping (Samuel and Larraza, 2015) but also greater lexical competition (Marian & Spivey, 2003). Further, they are known to perform more poorly than monolinguals in speech-in-noise comprehension (Rogers *et al.*, 2006). Given these consequences of bilingualism, the current study compares the intelligibility of foreign-accented speech for monolingual listeners and simultaneous bilingual listeners. Spanish-English bilinguals and English monolinguals performed a sentence repetition task for sentences produced by speakers of Mandarin-Chinese accented English with varying levels of proficiency in English. The competing hypotheses for this study are as follows: (1) because bilinguals' have a more flexible phonological-lexical mapping system, they will outperform the monolinguals in foreign-accented speech recognition; or, (2) greater lexical competition due to the bilinguals' L2 lexicon will result in poorer performance. The results will provide insight on bilingual sentence processing, and shed light on the shared and unique experiences of bilinguals and monolinguals.

**3pSCb9. Processing mixed talkers in noise suggests two mechanisms for perceptual adaption to speech.** Maya Saupe (Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, msaupe@bu.edu), Virginia Best (Speech, Lang. and Hearing Sci., Boston Univ., Boston, MA), Sung-Joo Lim (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., Boston, MA), Ja Young Choi (Boston Univ., Boston, MA), and Tyler K. Perrachione (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., Boston, MA)

Perceptual adaptation to a talker allows listeners to efficiently resolve inherent ambiguities present in the speech signal introduced by the lack of a one-to-one mapping between acoustic signals and intended phonemic categories across talkers. In ideal listening environments, preceding speech context enhances perceptual adaptation to a talker. However, little is known regarding how perceptual adaptation to speech occurs in more realistic listening environments with background noise. Here, we explored how talker variability and preceding speech context affect identification of phonetically confusable words in adverse listening conditions. With dependent variables of response time and threshold signal to noise ratio (SNR), our results showed that listeners were less accurate and slower in identifying mixed-talker speech compared to single-talker speech when target words were presented in multi-talker babble, and that preceding speech context enhanced word identification performance under noise both in single- and mixed-talker conditions. These results extend previous findings of perceptual adaptation to speech in quiet environments and suggest that two distinct mechanisms underlie perceptual adaptation to speech: rapid successful feedforward allocation of attention to salient talker-specific stimuli via auditory streaming, and an additional mechanism that preallocates cognitive resources to support processing of talker variability over longer time scales.

WEDNESDAY AFTERNOON, 9 DECEMBER 2020 — 1:50 P.M. TO 2:35 P.M.

## Session 3pSCc

## Speech Communication: Speech Articulation II (Poster Session)

Authors will be at their posters from 1:50 p.m. to 2:35 p.m.

### *Contributed Papers*

**3pSCc1. Using high-order derivatives of articulatory trajectories to identify gesture onsets.** Dan Cameron Burgdorf (Linguist., Cornell Univ., 28 Village Circle Apt. 2, Ithaca, NY 14850, dcb275@cornell.edu)

Articulatory trajectories are useful for measuring speech properties, but come with complications: a given trajectory may be influenced by multiple articulators and/or multiple overlapping gestures. Standard procedure is to identify landmarks from the trajectory and its velocity. I present a new procedure: higher order derivatives can be used to identify gesture onsets. High-order derivatives are generally associated with signal noise, and therefore not considered to be meaningful. More broadly, however, they reflect the jaggedness or un-smoothness of a signal, which can derive from sources other than random noise. I demonstrate that such jaggedness at gesture onsets is both predicted by the task dynamic model of speech production (Saltzman & Munhall 1989) and observed in real EMA data. Articulator trajectories generated with the Task Dynamics Application (TADA) Matlab implementation of this model (Nam & Goldstein 2004) show a pattern of increasing activity around changes in gestural activation and decreasing activity elsewhere as successive derivatives are taken. This was confirmed in EMA data on glides, demonstrating not only that the effect is real, but that it can be measurable above other noise. Activation noise can be isolated with targeted filtering, and may yield new insights into speech motor control.

**3pSCc2. Automatic classification of kinematic flap variants using ultrasound and optical flow.** Matthew Faytak (Univ. of California, Los Angeles, 3125 Campbell Hall Box 951543, Los Angeles, CA 90095, faytak@ucla.edu), Connor Mayer, Jennifer Kuo, G. Teixeira, and Z. L. Zhou (Linguist., Univ. of California, Los Angeles, Los Angeles, CA)

Study of the four covert kinematic variants of North American English flaps can provide insight into questions in speech motor control and articulation-acoustics relations [Derrick and Gick, *Can. J. Linguist.* **56**(3), 307–319 (2011)]. These variants are typically labeled by human annotators from ultrasound video, which is time-consuming and labor-intensive. In this study, we present an automatic classification method, taking as basis data optical flow fields over a series of ultrasound frames flanking the flap; fields are calculated using a method tailored to ultrasound video [Moisik *et al.*, *JIPA* **44**(1), 21–58 (2014)]. Two classifiers are compared: support vector machines, which learn an optimal linear separator between labeled class instances; and simple recurrent neural networks [Elman, *Cognit. Sci.* **14**(2), 179–211], which operate recursively over sequences of data, and whose state at any timepoint is calculated based only on the state at the preceding timepoint and the current input field. We train these classifiers on human-labeled flap tokens and test classification performance against a held-out subset of the labeled tokens. We go on to discuss the general applicability of this method for disambiguating covertly different lingual articulatory events.

**3pSCc3. Tongue root position during Mandarin Chinese stops.** Suzy Ahn (Dept. of Linguist., UCLA, 3125 Campbell Hall, Los Angeles, CA 90095, suzyahn17@gmail.com), Matthew Faytak (Dept. of Linguist., UCLA, Los Angeles, CA), and Harim Kwon (Dept. of English, George Mason Univ., Fairfax, VA)

Tongue root advancement is reported to be an adjustment that facilitates phonetic voicing during stop closure (Westbury 1983). In a recent ultrasound study, Ahn (2018) reports that the tongue root is more advanced during the American English "voiced" stops than the "voiceless" stops even without actual phonetic voicing during "voiced" stops, suggesting that tongue root adjustment might not be tied to phonetic voicing, but rather implementation of a more abstract contrast. The current ultrasound study compares tongue root position during Mandarin Chinese voiceless unaspirated stops /p, t, k/ and voiceless aspirated stops /$p^h$, $t^h$, $k^h$/. These two Mandarin categories are acoustically similar to English "voiced" and "voiceless" stops in phrase-initial position. To confirm the role of tongue root advancement in English "voiced" stops, this study investigates eight native Mandarin speakers' phrase-initial stops. If the tongue root configurations during Mandarin aspirated and unaspirated stops are similar to each other, unlike those during English "voiced" and "voiceless" stops, English and Mandarin arguably have different types of contrast. On the other hand, if Mandarin shows comparable results to English, it further confirms that the tongue root advancement may have a different motivation than merely facilitating phonetic voicing.

**3pSCc4. Articulation and identification of voiced stop consonants produced by acoustically driven vocal tract modulations.** 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)

A recently developed model of speech production [Story & Bunton, *JASA*, **146**(4), 2522–2528] was used to generate VCVs that were examined with regard to both articulation and identification of the consonant. In this model, an utterance is generated by specifying relative acoustic events along a time axis. These events consist of directional changes of the vocal tract resonance frequencies called resonance deflection patterns (RDPs) that, when associated with a temporal event function, are transformed via acoustic sensitivity functions, into time-varying modulations of the vocal tract shape. RDPs specifying /b/, /d/, and /g/ would typically be coded as $[-1\ -1\ -1]$, $[-1\ 1\ 1]$, and $[-1\ 1\ -1]$, respectively, indicating, from left to right, the targeted directional shift of the first, second, and third resonances of the vocal tract. In this study, two types of $V_1CV_2$ continua were constructed in three vowel contexts (/i, a, u/) by incrementing in small steps (1) the second resonance deflection from $-1$ to 1, and (2) the third resonance deflection from 1 to $-1$. The resulting time-varying vocal tract shapes emulate expected articulation patterns for the stop consonants, and a perceptual experiment indicated that listeners identify the consonants based on the polarity of RDP values.

**3pSCc5. Focusing on vertical larynx action dynamics.** Miran Oh (Linguist., Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, miranoh@usc.edu) and Yoonjeong Lee (Linguist., Univ. of California, Los Angeles, Los Angeles, CA)

Previous studies have shown a positive correlation between fundamental frequency and vertical larynx position. Additionally, Lee (2018) suggests the possibility that there is roughly one vertical larynx movement (VLM) per Accentual Phrase (AP) in Seoul Korean. Building on these findings, this study investigates the effect of prosodic focus on vertical larynx movements. Target sentences designed to have roughly four APs (e.g., $_{AP}$[*Joo-hyun*$_{SUBJ}$] $_{AP}$[*shabby garden field*] $_{AP}$[*six yards*$_{OBJ}$] $_{AP}$[*sold*$_{DECL}$], presented in Korean) were used to elicit a focus condition with focus on the initial word of the object phrase (e.g., *six*). Results from five Seoul Korean speakers indicate that quantifiable vertical larynx movements observed with real-time MRI for each sentence range from 3 to 6 movements, with 4 movements per sentence being most frequent. Sentences under focus had more instances of VLM per sentence than those without focus. Additionally, the displacement magnitudes of the vertical larynx gestures decrease over time through the sentence. Lastly, focused sentences have significantly greater VL displacement around the region of focus than the control. These observations on vertical larynx actions have implications for prosodic planning, downdrift, and pitch resetting, and future work is examining how VLMs align and correlate with Accentual Phrases and fundamental frequency. [Work supported by NIH.]

**3pSCc6. Ejective production mechanisms in English.** Lavinia Price (LMU Munich, Schellingstr. 3, München 80799, Germany, l.price@campus.lmu.de), Marianne Pouplier, and Philip Hoole (LMU Munich, Munich, Germany)

Ejective realization of stops seems to be increasing in many varieties of English, which has triggered a new debate on the basic mechanism of ejective production. While traditionally the glottalic initiation mechanism of ejectives is thought to require a closed glottis and active larynx raising, this has been questioned for the emergent English ejectives, and a pulmonic airstream mechanism has been suggested. In order to shed light on the possible production mechanism of English ejectives, we present rtMRI data acquired at 50fps from a corpus of 27 native British English and 18 native American English speakers. Auditory analysis of 20 BE-speakers from the corpus confirms that 26% of all word-final stops in target words are realized as ejectives. Our results so far agree with previous observations about the conditioning factors favoring ejective production. Ejectives occur predominantly in stressed position as allophones of word-final velar, less frequently of labial and coronal stops. Surprisingly, also phonemically voiced stops may be realized as ejectives. Importantly, the rtMRI data will allow us to measure larynx height and to observe the extent of active larynx movement involved in the initiation process in English ejectives. [Work supported by ERC Horizon 2020 grant Interaccent to J. Harrington.]

**3pSCc7. Effects of velopharyngeal size, airflow rate, and microphone placement on nasalance score.** Michael Rollins (Biomedical Eng., Univ. of Cincinnati, MSB 6303, 231 Albert Sabin Way, Cincinnati, OH 45267, rollinmk@mail.uc.edu) and Liran Oren (Dept. of Otolaryngology-Head and Neck Surgery, Univ. of Cincinnati, Cincinnati, OH)

Nasometers are designed to measure the nasalance score, which is the ratio of nasal acoustic energy to total (nasal plus oral) acoustic energy during production of sonorant phonemes. However, nasometer microphones can detect hydrodynamic pressure changes in addition to acoustic pressure changes. This can occur when a talker with incomplete velopharyngeal closure leaks airflow through the nasal cavity; these nasal emissions lead to increased aerodynamic flow impinging on the nasal microphone, which may significantly increase the nasalance score. In order to quantify the influence of nasal emissions on the nasalance score, airflow with no voiced component was passed through an [s]-postured airway model based on a subject-specific geometry. Nasometer microphones measured pressure fluctuations exterior to the nose and mouth, from which the nasalance score was calculated. Flow rate, size of the velopharyngeal opening, and microphone position (medial or lateral) were varied in a factorial experiment. It was found that in the presence of nasal emissions, placing the nasometer microphones laterally decreases the nasalance score. Furthermore, the nasalance score generally increases with increased velopharyngeal opening size. These results indicate that hydrodynamic pressure from nasal emissions can increase the nasalance score and that this influence could be mitigated via microphone placement.

**3pSCc8. A three-dimensional ultrasound investigation of the uvular rhotic: German /ʀ/ in six prevocalic contexts.** Tyler B. Kniess (Indiana Univ., 355 N Jordan Ave., Germanic Studies, Bloomington, IN 47401, tykniess@iu.edu)

The German rhotic is classified in the standard variety as a uvular trill, but has been shown to demonstrate considerable variation as a result of phonetic reduction processes. Most previous work on the German /ʀ/ has focused on its acoustic properties or its phonological patterning in particular prosodic positions, rarely addressing vowel quality and having limited means to investigate articulation. This paper presents data from a 3D/4D ultrasound investigation of word-initial /ʀ/ in six prevocalic contexts from two L1 German speakers. The vowel space was found to be consistent with Pouplier *et al.* (2004). The findings indicate that a groove along the posterior dorsum accompanies trilling and that /ʀ/ may be susceptible to coarticulatory effects from a following vowel. Specifically, tongue body height during /ʀ/ articulation can pattern with a following vowel, inhibiting trilling and thus resulting in a fricative, complementing the EMA findings of Schiller & Mooshammer (1995). This study therefore contributes to the body of literature concerning the diversity of rhotics as a class and the variability of the German rhotic in particular.

**3pSCc9. Voice quality and f0 correlates of Burmese tone in the standard and southern varieties.** Volker Dellwo (Phonet. and Speech Sci. Group, Universitaet Zurich, Plattenstrasse 54, Phonet. Lab., Zurich 8005, Switzerland, volker.dellwo@uzh.ch) and Mathias Jenny (Comparative Lang. Sci., Univ. of Zurich, Zurich, Switzerland)

Tones can be realized through fundamental frequency of oscillation ($f_o$) or voice quality correlates (open-quotient, jitter and shimmer). Here we investigated the acoustic correlates of tones across two varieties of Burmese (a) standard Burmese (STB) and (b) southern Burmese (SBM). Historically, Burmese likely had voice-quality contrasts, as suggested by the orthography of Old Burmese (12th century), while Modern Burmese has been described as tonal language. We were interested whether there are acoustic differences in the realisation of tone between the two variants. 20 speakers (10 STB, 10 SBM) produced the three phonological tone categories (neutral, creaky and heavy) either in carrier sentences in which the tones were in contrast ($N = 72$) or in sentences where the tones occurred without contrast ($N = 72$). Acoustic and electroglottographic signals were recorded of which acoustic correlates of pitch and voice quality ($f_o$, open-quotient, jitter, duration) were obtained. Results showed that STB predominantly used contrasts through voice quality correlates, and SBM used $f_o$. However, while SBM had more tone height differences, STB showed stronger slopes for two different tones. The variability of the tonal correlates between the two varieties may be relatet to possible external (contact, climatic) influences on tone systems.

WEDNESDAY AFTERNOON, 9 DECEMBER 2020 — 2:50 P.M. TO 3:35 P.M.

## Session 3pSCd

## Speech Communication: Speech Production in Second Language I (Poster Session)

Authors will be at their posters from 2:50 p.m. to 3:35 p.m.

### *Contributed Papers*

**3pSCd1. The durational modulation of repeated words in second language discourse by language experience.** Alif Silpachai (Appl. Linguist. and Technol., Iowa State Univ., 527 Farmhouse Ln., Ames, IA 50011-1054, alif@iastate.edu), Ivana Rehman, and John Levis (Appl. Linguist. and Technol., Iowa State Univ., Ames, IA)

Previous research suggests that repeated words in discourse are durationally shortened in comparison to the first mention, particularly when the words describe the same scene in a story. However, previous methods often relied on reading passages, which may be challenging to second language (L2) speakers or films, which require significant cultural comprehension. These methods may provide different findings from the accessibility of discourse referents in spontaneous speech using a picture narrative. This pilot study used a multi-scene picture narrative to elicit word reduction in spontaneous discourse. L2 English speakers with Korean, Chinese, Vietnamese, or Spanish as a first language narrated a story using a sequence of eight pictures/scenes about two strangers who collide and accidentally pick up each other's suitcase. Productions, when compared to native speakers of English, showed similar patterns of repeated word reduction. The results suggest that durations typically reset to full duration when words are repeated in different scenes, but they reduce within scenes. Results also suggest that the degree of second mention reductions vary modestly by first language. The results also show that a picture narrative was a promising method to elicit 2nd mention reductions in spontaneous speech and demonstrated durational sensitivity to scene changes.

**3pSCd2. Acoustic characteristics of vowel reduction in advanced Spanish-English bilinguals.** Jenna T. Conklin (Carleton College, One North College St., Goodsell Observatory, Northfield, MN 55057, jconklin@carleton.edu), Olga Dmitrieva, Ye-Jee Jung (Purdue Univ., West Lafayette, IN), and Weiyi Zhai (Carleton College, Northfield, MN)

Reduction of unstressed vowels is a well-known aspect of English phonology that is not present in Spanish, and the absence of such reduction contributes to accentedness in English for L1 Spanish speakers (Flege & Bohn, 1989). Previous studies have established that stress, coarticulatory effects of the previous consonant, and sociolinguistic factors can all lead to English vowel reduction in L1 and L2 speakers (Byers & Yavas, 2017). The current study investigates how lexical stress and sociolinguistic factors can affect vowel reduction in 10 American English monolinguals and 10 Spanish-English bilinguals from Colombia. A shadowing task provided acoustic measurements of formant frequencies for 60 target words in Spanish and English, encompassing five vowels from each language and two levels of stress. Factors such as language dominance, age of acquisition, linguistic attitudes, and length of residence were assessed through a questionnaire to examine their respective influences on the degree of vowel reduction shown by bilinguals relative to native monolinguals. Results address the degree of vowel reduction in L1 and L2 English and L1 Spanish, as impacted by linguistic and extra-linguistic factors.

**3pSCd3. Acoustic correlates of English clear speech produced by native Korean speakers.** Ye-Jee Jung (Purdue Univ., 215 Nimitz Dr. Apt. 4, West Lafayette, IN 47906, jung292@purdue.edu) and Olga Dmitrieva (Purdue Univ., West Lafayette, IN)

Monolingual clear speech is acoustically distinct from casual speech and can aid speech perception (Picheny *et al*., 1986; Smiljanic and Bradlow, 2005) but little is known about the acoustic properties of clear speech produced by non-native speakers. The current study aims to determine whether Korean non-native speakers of English modify their speech the same way native English speakers do when asked to speak clearly. A group of native Korean speakers residing in the United States (14 recorded to date) and a group of native speakers of English (9 recorded to date) were instructed to read a list of common English words (e.g., *beat*) using casual and clear speaking styles. The findings showed the effect of speaking style for every acoustic measurement: vowel duration, mean pitch, pitch range, positive and negative VOT, onset f0, and vowel space. With the exception of mean pitch, there were no significant interactions between group and speaking mode, indicating that Korean speakers implemented these acoustic parameters in L2 clear speech similarly to native speakers. Only for mean pitch, the degree of change associated with clear speech was greater for English speakers than for Korean speakers.

**3pSCd4. Monitoring self-produced speech variability in native and learned languages.** Sarah Bakst (Commun. Sci. and Disord., Univ. of Madison-Wisconsin, 1500 Highland Ave., Madison, WI 53705, sbakst@wisc.edu) and Caroline A. Niziolek (Commun. Sci. and Disord., Univ. of Wisconsin–Madison, Madison, WI)

Learning to produce new contrasts in a second language (L2) is challenging because learners have a less refined model of their new speech targets. Native speakers will correct subphonemic variability in their own speech, where greater variance triggers a greater degree of correction. Here we consider English (L1) speakers learning to produce front-rounded vowels as they acquire French or German. In one study, speakers uttered monosyllabic English and L2 words while simultaneously receiving different levels of masking noise to determine how access to auditory feedback information impacts speakers' ability to make online adjustments to self-produced speech. There was greater acoustic variability at both onset and midpoint of vowels in L2 than in L1, regardless of noise condition, consistent with a differential precision in executing motor programs in the two languages. In a second study, speakers received altered auditory feedback. On one-third of trials, the vowel formants of produced words were shifted up or down, creating the perception of an error. Compensation onset was earlier in L2, suggesting a stronger reliance on feedback than feedforward information in L2 production. In our ongoing studies, we investigate whether perceptual acuity in F2 may underlie these differences.

**3pSCd5. Real-time visual acoustic feedback for non-native vowel production training.** Ivana Rehman (English, Iowa State Univ., 4716 Hutchison St., Apt. 2, Ames, IA 50014, ilucic@iastate.edu) and Anurag Das (Texas A&M Univ., College Station, TX)

Computer-Assisted Pronunciation Training (CAPT) software which include visual feedback have shown potential for second language (L2) learning (Mehta & Katz, 2018). However, real-time formant visualization has yet to be used for L2 learners' production of speech sounds. This project investigates the effects of real-time formant visualization on L2 vowel production training. First, a vowel visualization system is presented. In contrast with other similar tools which use a regression approach (Frostel *et al*., 2011), this real-time formant (F1 and F2) extraction is performed using the Parselmouth library (Jadoul *et al*., 2018), which then uses Praat's formant extraction algorithm (Gray & Wong, 1980) to ensure accuracy. Then, the results from a preliminary user study are reported. L2 learners (n = 5) participated in six 30-min training sessions, during which they used the system to practice their production of eight vowels. Pronunciation accuracy was analyzed by calculating Mahalanobis distances between participants' productions (produced in pre-test and post-test) and native exemplars. The analysis of this preliminary dataset showed that the use of real-time visual acoustic feedback resulted in L2 vowel production improvement, which suggests that this system could, therefore, be used as a pedagogical tool for L2 learners.

**3pSCd6. Bilingual language processing: Interactions between lexical retrieval and vowel production.** Maria F. Gavino (Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, mariagavino@u.northwestern.edu) and Matthew Goldrick (Northwestern Univ., Evanston, IL)

What is the relationship between lexical retrieval and phonetic production in bilingual language processing? Various factors related to bilingual language processing affect bilingual's selection of context-appropriate words and speech sounds. One factor is whether bilinguals are using one (single context) or both (mixed context) languages. Increased language selection difficulty in mixed contexts (especially when the previous word is in a different language than the target word; i.e., switch context) slow down retrieval and increase accentedness. Another factor is whether a word has two highly distinct forms (non-cognates) or highly similar forms (cognates) for a concept in both languages. Increased cross language activation for cognates facilitate retrieval, but increase accentedness. In this project, 18 Spanish-English bilinguals named pictures of cognate and non-cognate words in single and mixed contexts in Spanish and English. Reaction time and vowel formants were analyzed. Results show that there are cognate facilitation effects, mixing, and switching costs for retrieval, but only consistent mixing costs for accentedness. The dissociation between these effects during lexical retrieval and phonetic production suggests continuing interactions between them after the initiation of the response.



**3pSCd7. Childhood language exposure: Does early experience affect sound perception and production in speakers with reduced language proficiency?** Rawan Hanini (Commun. Arts, Sci., & Disord., CUNY Brooklyn College, 2900 Bedford Ave., Brooklyn, NY 11210, rawan.hanini26@bcmail.cuny.edu), Anwar Alkhudidi (Linguist., Univ. of Western ON, London, ON, Canada), and Yasman Rafat (Lang. and Cultures, Univ. of Western ON, London, ON, Canada)

Early language exposure is crucial for acquiring native mastery of the phonology of a language (Fromkin 1974, Flege 1987, Antoniou *et al*. 2015). It is not clear, however, whether early language exposure has lasting benefits when the quantity and quality of speaking drop dramatically after childhood (Oh *et al*. 2003). In this study, we investigate the production of Arabic speech sounds in 20 native speakers (who heard and spoke Arabic during childhood regularly), 20 childhood speakers (who heard Arabic regularly during childhood but did not speak it regularly afterwards), and 20 novice speakers (did not have any exposure to Arabic during childhood but are currently enrolled in Arabic language classes). The experiment includes a language proficiency test and a phoneme production and perception task. The target sounds are geminate consonants. The acoustic analysis consists of manual alignment of each consonant, and extracting their duration and acoustic information (voicing, aspiration, and formant values). The findings shed more light on whether early language experience has measurable long-term benefits for an individual's phonetic and phonological skill even if the language experience diminishes over time. In addition, we gain further insight into the role played by universal markedness factors in L2 acquisition (Davidson, 2011).

**3pSCd8. Phonetic interaction between late Japanese-English bilinguals' L1 and L2 vowels—A longitudinal study.** Chikako Takahashi (Dept. of Linguist., SUNY at Stony Brook, Stony Brook, NY 11794-4376, chikako.takahashi@stonybrook.edu)

We investigated late bilinguals' (N = 31) change in their production and perception of the L1 Japanese vowel /i/ and its phonetically associated L2 vowel /i/ and /ɪ/ over one year. The perceptual boundaries between both English /i/-/ɪ/ and Japanese /i/-/e/ were examined through identification tasks. Participants' production of the high front vowels was also analyzed. More than half of the participants exhibited evidence of separating English and Japanese high front vowels from each other in production. Participants who seemed to use similar F1-F2 values for these vowels still tended to separate them. The perception task results showed a perceptual boundary shift in bilinguals' L1 Japanese after one year. That is, they identified more tokens as /i/ than /e/ in /i/-/e/ continuum. This phenomenon was also observed when comparing bilinguals to Japanese monolingual controls. We argue that this change in L1 perception stems from learning a new L2 phonetic category /ɪ/ and associating it with a similar L1 /i/. Individual differences in bilinguals' relative language use and exposure were also examined as potential factors driving changes in perception and production of Japanese/English high front vowels.

**3pSCd9. Second-language learners of French do not acquire habitual pitch values.** Alicia Mason (Dept. of Linguist., New York Univ., 10 Washington Pl., New York, NY 10003, aem782@nyu.edu)

Järvinen *et al*. (2013) find that native Finnish speakers have higher pitch in English, suggesting that habitual pitch can be learned during second language acquisition. The mean F0 of French may be higher than that of English (Pépiot 2014), and this study investigates whether native speakers of American English who are L2 learners of French show different mean F0 in their second language compared to their first, and whether this is affected by L2 proficiency. The study compared the average F0 in sentences spoken by 6 female L2 French speakers (3 high proficiency, 3 low proficiency) to their English F0, and to the F0 of 6 female L1 French speakers. The L2 French speakers' F0 showed no significant difference from their L1 English, but L1 French F0 was significantly higher than both English and L2 French ($p < 0.01$ in both cases). There was also no significant difference in F0 between the high-proficiency and low-proficiency L2 French speakers. This indicates that English-speaking L2 learners do not acquire the higher habitual pitch values of French, even as proficiency increases. A comparison with Järvinen et al's findings suggests that adaptation to L2 pitch may not be automatic and may vary by language.

**3pSCd10. Voice onset time in monolingual and code switching modes in Lebanese Arabic and English.** Niamh Kelly (American Univ. of Beirut, P.O.Box 11-0236, Beirut 1107 2020, Lebanon, nk114@aub.edu.lb), Mia El Houry (American Univ. of Beirut, Beirut, Lebanon), and Farah Ghamloush (American Univ. of Beirut, Beirut, Lebanon)

The effect of speaking mode and language in speakers of Lebanese Arabic and English was examined. Voice onset time (VOT) was measured for voiced and voiceless stops in both languages, when speakers were in monolingual mode or code-switching mode. Nine speakers who were highly proficient in English, and used to code-switching, produced 534 tokens in an image-labelling experiment (Olson, 2013; Mayr *et al*., 2018). Research has found that speakers may produce the same sound differently depending on whether they are in monolingual or bilingual mode (e.g., Olson, 2013; Simonet, 2014; Amengual, 2018). Linear regressions found no significant difference in VOT between conditions (monolingual versus code-switching) or between languages (Arabic versus English). The only significant predictor of VOT was voicing of the stops. Average VOT for voiced and voiceless stops in monolingual Arabic: -59msec & 44msec, monolingual English: -55 ms and 48 ms, code-switched Arabic: -63 ms and 45 ms, and code-switched English: -56 ms and 47 ms. These results indicate voicing during the closure for voiced stops and some aspiration for voiceless stops, and show that speakers do not have separate stop categories for the two languages. Current research is examining monolingual Arabic speakers to determine whether the bilingual speakers have L2 transfer from English aspiration.

WEDNESDAY AFTERNOON, 9 DECEMBER 2020 — 3:35 P.M. TO 4:20 P.M.

## Session 3pSCe

## Speech Communication: Speech Production in Second Language II (Poster Session)

Authors will be at their posters from 3:35 p.m. to 4:20 p.m.

### *Contributed Papers*

**3pSCe1. Acoustic measures of glottalization in native and nonnative English connected speech.** Michael Fang (Dept. of Linguist., Boston Univ., 621 Commonwealth Ave., Boston, MA 02215, mfmfmf@bu.edu) and Charles B. Chang (Dept. of Linguist., Boston Univ., Boston, MA)

Using perceptual coding methods, previous research on English connected speech by late-onset second language speakers (L2ers) found that L2ers produced unstressed vowel-to-vowel word junctures (e.g., *go héad*) as "delinked" (i.e., glottalized) at significantly higher rates than native speakers (L1ers), which could not be explained by cross-linguistic influence (CLI) from the L2ers' native language (L1). In the present study, we examined whether L2ers differed from L1ers also in degree of glottalization. Comparing high-proficiency L2ers from an L1 Mandarin background ($N = 5$) with L1ers ($N = 5$), we collected four measures of spectral tilt ($H_1$–$H_2$, $H_1$–$A_1$, $H_1$–$A_2$, $H_1$–$A_3$) at vowel-to-vowel word junctures spontaneously produced in interview speech ($N = 615$ for L1ers, $N = 841$ for L2ers). Results showed that although junctures coded as glottalized tended to show more negative $H_1$–$H_2$ values (indicative of creaky phonation) than did junctures coded as not glottalized, there were no significant differences between L1ers and L2ers in any of the spectral tilt measures examined, for either glottalized or non-glottalized junctures. These findings thus suggest that although high-proficiency L2ers may glottalize vowel-to-vowel junctures at higher rates than L1ers, when they do glottalize they do so to a similar degree as L1ers, with no apparent CLI from the L1.

**3pSCe2. The relation between category compactness and L2 VOT learning.** Marie K. Huffman (Dept. Linguist., Stony Brook Univ., SBS S 201, Stony Brook, NY 11794-4376, marie.huffman@stonybrook.edu) and Katharina Schuhmann (Dept. of Germanic & Slavic Lang. & Literatures, and Linguist. Program, Penn State Univ., State College, PA)

L1 and L2 category compactness have been reported to correlate with L2 vowel pronunciation accuracy in advanced intermediate learners (Kartushina and Frauenfelder, 2014). However, it is not clear whether category compactness is an effect of L2 learning, or whether it is an individual characteristic that might predict aspects of L2 learning. Holliday (2015) suggests that more variable VOT categories early in L2 learning may facilitate acquisition of new L2 VOT patterns. We tested relationships between L1 and L2 category compactness and L2 pronunciation progress for voiceless stops, in ten L1 English-L2 Spanish learners over the course of one semester of early Spanish courses. We found that lower L2 Spanish VOT correlated with higher voiceless VOT standard deviation at the end of the semester for all but the absolute beginners ($R2 = 0.6256$, $F(1,5) = 37.75$, $p = 0.004$). Thus, for early learners, as production improves, L2 categories may expand. In addition, L1 English voiceless stop VOT category compactness at beginning of term correlated with L2 Spanish VOT accuracy at end of term for all learners ($R2 = 0.6798$, $F(1,9) = 17.0$, $p = 0.003$), suggesting that L1 category compactness may reflect individual properties that influence progress in L2 learning.

**3pSCe3. Voicing contrasts in the stops of Indian English produced by Assamese speakers.** Caroline Wiltshire (Linguist., Univ. of Florida, Box 115454, Gainesville, FL 32611-5454, wiltshir@ufl.edu) and Priyankoo Sarmah (Indian Inst. of Technol. Guwahati, Guwahati, Assam, India)

Speakers of most American and British varieties of English contrast word-initial stops using aspiration, with long-lag Voice Onset Time (VOT) for /ptk/ and short-lag or lead voicing for /bdg/ (Docherty 1992, Chodroff & Wilson 2017, etc.). However, Indian English (IndE) speakers of Hindi and Telugu backgrounds living in the US produce a short-lag versus lead voicing contrast instead (Davis & Beckman 1983, Sirsa & Redford 2013). We recorded 10 L1 Assamese speakers residing in India reading English word-lists, all bilinguals who attended English-medium schools starting at age 4.3 ($\pm$1.3), young enough to acquire the target English of their community (Flege 1991). We are measuring the VOT of ten tokens each of /ptkbdg/ in word-initial position. Results thus far support the interpretation of the contrast in IndE as voicing not aspiration, as unaspirated voiceless stops versus pre-voiced stops. We also measure two expected consequences of using [voice] for the contrast: full voicing of sonorant consonants after voiceless obstruents in onset clusters and full (>75% of closure), rather than passive, intervocalic voicing. Though neither occurs in aspirating languages (Beckman *et al.* 2013), we find both in the IndE of Assamese speakers, suggesting IndE has developed a distinct contrast from AmE/BrE.

**3pSCe4. An examination of articulatory skill in monolinguals and multilinguals: A tongue twister experiment.** Crystal Gilbert (Commun. Sci. and Disord., Brooklyn College, 642 Cleveland St., Brooklyn, NY, icrystalgilbert@gmail.com), Beckie Dugaillard (Speech Lang., Lehman College, Westchester, NY), Marianna Krivoshaev (Linguist. & Commun. Disord., Queens College, Brooklyn, NY), and Laura Spinu (Communications & Performing Arts, Kingsborough Community College, Brooklyn, NY)

The main goal of this study is to determine whether articulatory differences exist between monolinguals ($n = 19$) and bilinguals ($n = 21$) through the analysis of tongue-twister production, following Goldrick & Blumstein (2006). The latter were divided into early bilinguals (consistent exposure to both of their languages before age 5, $n = 8$), mid bilinguals (exposure to L2 between 5 and 10 years, $n = 5$), late bilinguals (L2 exposure between 10 and 15 years, $n = 3$) and trilinguals ($n = 5$).The stimuli comprised 64 sequences that each contained four syllables (e.g., kif tif tif kif) and had to be repeated three times to a beat of a metronome (150 beats per minute). The recordings were rated by a trained listener, who gave a score of 1 for each accurately produced onset and coda consonant, and a score of 0 otherwise. In addition, a subset of the sounds produced were manually aligned in Praat to validate the raters' scoring and determine whether speakers also differed in less perceptible aspects of speech. The results show that while there are no overall differences in accuracy between the two groups (monolingual versus multilingual), bilinguals who were first exposed to their second language later in life (between 5 and 15) exhibit a statistically significant advantage in the articulation of tongue-twisters (Dugaillard & Spinu, 2019). Our findings underscore the importance of directly measuring bilingual language proficiency and incorporating this information to experimental design (DelMaschio & Abutalebi 2018, Sulpizio *et al.* 2019).

**3pSCe5. Voice quality differences between American English speakers and Korean learners of English.** Hahn Koo (San Jose State Univ., One Washington Square, San Jose, CA 95192-0093, hahn.koo@sjsu.edu)

Various features have been examined as markers of foreign accent including segment-level errors, deviation in subsegmental acoustic properties, as well as prosodic patterns such as intonation, rhythm, and speaking rate. Features that characterize voice quality may also be useful but they have been investigated less extensively in the literature. The present study explores the utility of voice quality features by analyzing and comparing speech samples from American English speakers and Korean learners of English documented in the Wildcat Corpus of Native- and Foreign-Accented English. Measures of jitter, shimmer, and harmonics-to-noise ratio (HNR) were extracted from vowels and compared between the two speaker groups. Overall, Korean learners produced English vowels with less jitter and shimmer as well as higher HNRs than English speakers. The pattern was consistently found in both scripted and unscripted speech and was more salient among male speakers than female speakers. Furthermore, the same trend, although weaker, was also found when the Korean and English speakers produced vowels in their respective L1s while reading comparable scripts. However, the Korean speakers showed more shimmer while reading scripts in Korean (L1) than English (L2).

**3pSCe6. The role of passage length on acoustic voice variability in bilingual speech.** Khia A. Johnson (Linguist., Univ. of BC, 2613 West Mall, Vancouver, BC V6T 1Z2, Canada, khia.johnson@ubc.ca) and Molly Babel (Linguist., Univ. of BC, Vancouver, BC, Canada)

An individual's voice is determined in part by the limitations of their anatomy and physiology, in addition to language-specific phonological and phonetic structure. When a bilingual switches between languages, how much do they change their voice? Previous work using a corpus of spontaneous speech from early Cantonese-English bilinguals found surprisingly little variability across individuals' languages [Johnson *et al.*, Proc. of Interspeech (2020)] compared to earlier research on across-talker acoustic voice variability [Lee *et al.*, JASA (2019)]. A crucial difference between these two studies, however, is passage length. A longer passage (e.g., 30 min) potentially allows for a more stable structure to emerge in a principal components analysis, while a shorter sample (e.g., 2 min or less) may instead be subject to ephemeral variation, and potentially misrepresent the overall variability of a voice. Building on Johnson *et al.* (2020), the present study asks: to what extent does passage length impact the results of principal components and canonical redundancy analyses designed to elucidate within-talker (across languages) and across-talker (within language) idiosyncratic variation? These results are important for theories of talker recognition, identification, and discrimination, in addition to improving understanding of talker-specific acoustic-phonetic variation.

**3pSCe7. Comparing Mandarin-English bilinguals and English monolinguals on executive function tasks.** Hiu Tung Gloria Lam (Federal Way Public Schools, 1715 NE Columbia Rd., Seattle, WA 98195, htglam@outlook.com), Tian Zhao, and Patricia K. Kuhl (Inst. for Learning and Brain Sci., Univ. of Washington, Seattle, WA)

Previous studies report mixed evidence regarding bilinguals' cognitive advantage in inhibition and other executive function (EF) tasks compared to monolinguals. This study aims to replicate previous research on Mandarin-English bilinguals. Forty-four participants (22 English monolinguals & 22 Mandarin-English bilinguals) were recruited to complete four EF tasks: Simon, Stroop, Letter-Number Switching and Color-Shape Switching (Cued). The Simon and Stroop tests target inhibition by examining responses to trials with compatible versus incompatible information, and the

switching task targets attention shifting skills by examining responses to trials that were repetitive versus changing. Reaction time and response accuracy were the dependent measures. In the Simon task, bilinguals responded significantly slower overall than monolinguals, although there was a significantly larger difference between compatible versus incompatible trials in monolinguals. The Stroop task revealed no group differences or interaction effects. For the switching tasks, bilinguals had a significantly higher percentage accuracy overall than monolinguals in the Letter-Number Switching task with no significant group differences found in reaction time. No significant group differences were found in both measures for the Color-Shape Switching (Cued) task. Overall, we report weak evidence of enhanced EF skills in Mandarin-English bilinguals and the results will be discussed in relation to previous studies.

**3pSCe8. The analysis of phonetic fine-tuning for less frequent nasalization developed in native Korean speakers' English production.** Wooji Park (English Lang. & Lit., Yonsei Univ., 50, Yonsei-ro, Seodaemun-gu, Seoul 03722, South Korea, woojipark@yonsei.ac.kr)

In Korean, a preceding stop sound is obligatorily nasalized by an immediately following nasal stop. Interestingly, this nasalization also occurs in English but less often than in Korean. Based on the difference in nasalization between English and Korean, this study investigated the degree and frequency of nasalization made by native English and Korean speakers when they produce the English stops /b, d, g, p, t, k/ before the nasals sounds /m, n/ in sentences. Twenty-five participants performed the task to speak 12 target items mixed with filler sentences. The results revealed that the native English speakers rarely nasalized the stops before the nasal sounds. On the contrary, the native Korean speakers generally nasalized them while each individual showed the different degree and frequency of nasalization from no nasalization to complete nasalization. It was also discovered that the voiced stops were more frequently nasalized than the voiceless stops across the Korean participants. Besides, the higher their English proficiency, the less likely nasalization appeared. It signifies that phonetic fine-tuning for less frequent nasalization in English develops among Korean speakers as they become more proficient in English.

**3pSCe9. Multilingualism and acoustic correlates of breathiness and tone in Kuy.** Raksit T. Lau-Preechathammarach (Univ. of California Berkeley, 1203 Dwinelle Hall #2650, Berkeley, CA 94720-2650, raksit@berkeley.edu), Melody J. Tran (Univ. of California Berkeley, Berkeley, CA), Stacey H. Vu (UC Berkeley, Berkeley, CA), and Cynthia Zhong (Linguist., Univ. of California, Berkeley, Berkeley, CA)

Previous work on tonogenesis has focused primarily on intergenerational differences in cue weighting (Kang 2014, Coetzee *et al.* 2018) and less so on other sociolinguistic and language usage factors (though see Brunelle 2009). The current study explores the relationship of these variables with acoustic measures of voice quality (H1*-H2*, H1*-A1*, H1*-A2*, H1*-A3*, CPP, F1, f0) in a modal-breathy distinction in 63 speakers of Kuy, a minority language of Thailand. Kuy speakers are historically quadrilingual in Kuy, Thai, Lao, and Khmer, but recent centralization of Thailand has led to greater Thai usage. Because Thai is tonal and lacks a voice quality distinction, speakers who use Thai more are hypothesized to weigh f0 more heavily and other breathiness measures less heavily. We find that f0 differences are significantly larger for speakers who use Thai more, while H1*-H2*, H1*-A1*, H1*-A2*, and H1*-A3* differences are slightly, but significantly, smaller. F1 shows no effect. Unexpectedly, CPP differences are also significantly larger for speakers who use Thai more. The results suggest that increasing Thai bilingualism plays a role in shifting voice quality cue weights and, in particular, in increasing f0 cue weights, providing one mechanism by which bilingualism in a tonal language may help catalyze tonogenesis.

**3pSCe10. Tone differentiation as a means for assessing non-native imitation of Thai tones by Mandarin speakers.** Juqiang Chen (Western Sydney Univ., The MARCS Inst. for Brain Behaviour and Development, Western Sydney University Locked Bag 1797 Penrith, Sydney, New South Wales 2751, Australia, j.chen2@westernsydney.edu.au), Catherine T. Best, and Mark Antoniou (Western Sydney Univ., The MARCS Inst. for Brain Behaviour and Development, Sydney, New South Wales, Australia)

Non-native tone production and imitation have been found to be phonetically deviant from native production for some discrete measures. However, it remains unresolved whether non-native imitation differs from native production in terms of the differentiation of tones in acoustic tone space. 32 native Mandarin speakers who had no experience with Thai imitated five Thai tones, and each participant produced 160 tokens in total under differing memory load and stimulus variability conditions to determine effects of cognitive demands. We calculated two tone differentiation indices (i.e., Index 1: tone differentiation within the tonal space; Index 2: differentiation among tones, both as in Barry & Blamey, 2004) based on F0 onset and F0 offset for Thai tones and the non-native imitations of these Thai tones by Mandarin imitators. There was a significant memory load by vowel variability interaction for Index 1 and a main effect of talker variability and a three-way interaction (memory load "talker variability" vowel variability) for Index 2, suggesting that tone differentiation is affected by cognitive factors. Nonetheless, non-native tone imitations were not significantly different from native productions on either index, indicating that non-native imitation resembles native production in terms of tone differentiation in an onset-offset F0 space.

## Session 3pSPa

# Signal Processing in Acoustics, Underwater Acoustics, Computational Acoustics, Acoustical Oceanography, and Animal Bioacoustics: Machine Learning in Acoustics I

Erin M. Fischell, Cochair
*Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, 266 Woods Hole Rd., MS 11, Woods Hole, MA 02543*

Daniel Plotnick, Cochair
*Penn State, State College, PA 16804*

Wu-Jung Lee, Cochair
*Applied Physics Laboratory, University of Washington, 1013 NE 40th St., Seattle, WA 98105*

**Chair's Introduction—1:05**

## *Invited Papers*

**1:10**

**3pSPa1. Alternative representations and object classification of circular synthetic aperture in-air acoustic data.** J. Daniel Park (Appl. Res. Lab., The Penn State Univ., P. O. Box 30, State College, PA 16804, jdp971@psu.edu), Thomas E. Blanford, Daniel C. Brown, and Daniel Plotnick (The Penn State Univ., State College, PA)

Synthetic aperture sonar imagery is typically generated using data collected with unmanned underwater vehicles. The prohibitive cost of collecting underwater data and the need for well-controlled factors such as collection geometry and object configuration has provided the motivation for devising a benchtop in-air circular acoustic data collection framework. This set-up makes it practically feasible to explore a multitude of parameters that are not as feasible with underwater measurement scenarios, including waveform type, object shapes and material. It is also practically feasible to explore various representations of the collected acoustic data that help better emphasize different aspects of the information embedded in the acoustic signal, which various machine learning algorithms can utilize. Signal processing and feature organization are critical to improving performance of machine learning algorithms. For example, geometric scattering response of objects is well-represented in spatial imagery with sharp contrast of pixel intensity between the object and surrounding environment, while spatial spectrum of the complex SAS image better represents the aspect-dependent spectral response of the object that help discriminate objects of the same shape, but with different material. We will discuss the relationship between the choice of representation and discriminatory information with illustrative classification problems.

**1:30**

**3pSPa2. Toward explainable convolutional neural network classifiers with acoustic-color sonar data.** David Williams (NATO STO Ctr. for Maritime Res. and Experimentation (CMRE), Viale San Bartolomeo 400, La Spezia, SP 19126, Italy, david.williams@cmre.nato.int) and Aubrey España (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Convolutional neural networks (CNNs) have achieved state-of-the-art performance on a wide range of object classification tasks, including in the underwater domain. But one frequent criticism of the technique is that the features uncovered by the networks and used to make predictions are poorly understood. In this work, we develop CNNs whose predictions are *explainable* in the sense that they can be tied directly to physical properties of the objects under study. To achieve this, we consider limited-scope experiments in which the two classes of objects in a binary classification task differ by a single physical attribute. Importantly, to avoid overfitting and improve generalizability, the CNNs are designed with limited capacities and relatively small numbers of parameters compared to what is common in the literature. We then link the discriminatory clues discovered by the CNNs back to principled, understandable wave phenomena associated with the objects. The data used to conduct the experiments are measured low-frequency sonar data in the form of acoustic-color plots, which show target strength as a function of object aspect and frequency.

3p WED. PM

*Contributed Paper*

1:50

**3pSPa3. Categorization of broadband spectra of mesopelagic targets using model-generated training data.** Emma D. Cotter (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 86 Water St., MS 11, Woods Hole, MA 02543, ecotter@whoi.edu) and Andone C. Lavery (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Recent studies using acoustic techniques suggest that mesopelagic biomass may be an order of magnitude higher than previously predicted by trawling, which has spurred increased interest in commercial exploitation of the mesopelagic zone. However, because of the limited understanding of species distributions in the mesopelagic, there is significant uncertainty surrounding these estimates. Here, we deploy a broadband (25–40 kHz) echosounder to mesopelagic depths to measure the target strengths of individual organisms. Automatic target detection and tracking are used to isolate individuals from the recorded data. Physics-based scattering models are then employed to generate a training data set with five broad classes: gas-bearing organisms that are observed above, below, or at their resonant peak, and fluid-like organisms that are observed either in the Rayleigh or Geometric scattering regimes. These training data are used to train a machine learning model to categorize the measured target spectra. The classification results agree well with human annotation and provide insight into both the distribution of organisms in the mesopelagic and the types of organisms that contribute most significantly to volume backscatter measured by shipboard echosounders.

WEDNESDAY AFTERNOON, 9 DECEMBER 2020 — 2:50 P.M. TO 4:20 P.M.

## Session 3pSPb

## Signal Processing in Acoustics, Underwater Acoustics, Computational Acoustics, Acoustical Oceanography, and Animal Bioacoustics: Machine Learning in Acoustics II

Erin M. Fischell, Cochair
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Daniel Plotnick, Cochair
*Penn State, Penn State University, State College, PA 16804*

Wu-Jung Lee, Cochair
*Applied Physics Laboratory, University of Washington, 1013 NE 40th St., Seattle, WA 98105*

**Chair's Introduction—2:50**

*Invited Papers*

2:55

**3pSPb1. Semi-supervised source localization in reverberant environments using deep generative modeling.** Michael J. Bianco (Marine Physical Lab., Univ. of California San Diego, Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92037, mbianco@ucsd.edu), Sharon Gannot (Faculty of Eng., Bar-Ilan Univ., Ramat-Gan, Israel), Efren Fernandez-Grande (Elec. Eng., Tech. Univ. of Denmark, Copenhagen, Denmark), and Peter Gerstoft (Marine Physical Lab., Univ. of California San Diego, Scripps Inst. of Oceanogr., La Jolla, CA)

We present a method for acoustic source localization in reverberant environments based on semi-supervised machine learning (ML) with deep generative models. Source localization in the presence of reverberation remains a major challenge, which recent ML techniques have shown promise in addressing. Despite often large data volumes, the number of labels available for supervised learning in reverberant environments is usually small. In semi-supervised learning, ML systems are trained using many examples with only few labels, with the goal of exploiting the natural structure of the data. We use variational autoencoders (VAEs), which are generative neural networks (NNs) that rely on explicit probabilistic representations, to model the latent distribution of reverberant acoustic data. VAEs consist of an encoder NN, which maps complex input distributions to simpler parametric distributions (e.g., Gaussian), and a decoder NN which approximates the training examples. The VAE is trained to generate the phase of relative transfer functions (RTFs) between two microphones in reverberant environments, in parallel with a DOA classifier, on both labeled and unlabeled RTF samples. The performance this VAE-based approach is compared with conventional and ML-based localization in simulated and real-world scenarios.

3:15

**3pSPb2. Bayesian framework for three dimensional, near field source localization using spherical harmonics.** Thomas Metzger (School of Architecture, Rensselaer Polytechnic Inst., 121 4th St., Troy, NY 12180-3912, tommy.r.metzger@gmail.com) and Ning Xiang (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

Localization of sound sources relative to a receiver is of great interest in many acoustical applications. Within the near field of a spherical array, spherical wave incidence can be assumed and both direction of arrival and radial distance information can be extrapolated. This work presents a two-tiered Bayesian framework, using spherical harmonics, through which the directions of arrival and radial distance of a single, or multiple sources in the near field relative to a spherical microphone array receiver can be estimated. The first tier estimates the number of sources present in the sound field, while the second tier estimates the direction of arrival and radial distance parameters. Two models for spherical wave incidence on a spherical array, both based on spherical harmonic beamforming techniques, are formulated and analyzed. This work presents analysis of the Bayesian framework and both models to demonstrate the feasibility of model-based Bayesian inference for three dimensional source localization of one or more sources in complex noise environments.

3:35

**3pSPb3. Effects of medium heterogeneities on direction of arrival estimation.** Gaultier Real (DGA Naval Systems, Toulon, France), George Sklivanitis (Florida Atlantic Univ., 777 Glades Rd., Boca Raton, FL 33431, gsklivanitis@fau.edu), Kay Gemba (Acoust. Div., Code 7162, US Naval Res. Lab., Washington DC, DC), and Dimitris A. Pados (Florida Atlantic Univ., Boca Raton, FL)

A scaled experiment in a water tank was conducted to study effects of medium heterogeneities on underwater acoustic propagation while controlling the so-called fluctuations. An ultrasonic wave is transmitted (a pulse train at center frequency $f = 2.25$ MHz) throughout an acoustic lens presenting a plane input face and a randomly rough output face inducing spatial fluctuations of the sound pressure field. The roughness statistics are tuned so that the fluctuating pressure field presents features comparable to what can be observed in the case of acoustic waves propagating through a fluctuating ocean. Generally speaking, disturbances can include path propagation delays, multipath dependent Doppler spreading, scattering from the sea surface or bottom, and coherence loss over the array aperture arising from dynamic wave-front fluctuations due to internal waves. This last topic is the main focus of our study. Signal snapshots received at a virtual, vertical line array of 64 sensors may be corrupted by impulsive (heavy-tailed) additive noise and high resolution outlier-resistant arrival angle identification is desirable. Recent advances in robust techniques such as Sparse Bayesian Learning and L1 -norm Principal Component Analysis motivate this work. We compare the performance of these innovative techniques to the classical MUSIC and Bartlett processors. [This work is supported by DGA, NSF Grant CNS1753406, ONR.]

## *Contributed Paper*

3:55

**3pSPb4. Automated matching of measured long-range acoustic arrivals from autonomous gliders with acoustic predictions.** Cristian E. Graupe (Ocean Eng., Univ. of Rhode Island, 30 Fish Rd., URI Narragansett Bay Campus Rm 13-14, Narragansett, RI 02882, graupec@my.uri.edu), Lora Van Uffelen (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of San Diego, San Diego, CA), Bruce M. Howe (Dept. of Ocean and Resources Eng., Univ. of Hawaii, Honolulu, HI), and Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA)

The acoustic arrival matching localization method estimates the range of a mobile receiver from a fixed broadband acoustic source by matching the arrival with an acoustic prediction of the received arrival at a known range, calculating an offset from the known range, and combining the estimates of range from multiple sources using least-squares to estimate subsurface position. This is a labor-intensive task performed in post-processing, after recovery of the mobile platform. Automation of this matching is not straightforward due to the presence of spurious arrivals and other variations in the measured arrivals compared with predictions. Automated methods for estimating the range errors and performing ray identification for specific peak arrivals were developed using both programmatic and machine learning approaches. These methods take advantage of the temporal structure of the measured arrivals and simultaneously estimate range error and resolve measured peaks by comparing against a reference acoustic propagation model that describes the progression of the acoustic arrival structure in depth, range, and reduced travel time. The programmatic and machine learning approaches will be tested on range dependent ray simulations and evaluated in the context of a dataset collected using Seagliders during the North Pacific Acoustic Laboratory (NPAL) Philippine Sea Experiment.

## Plenary Session and Awards Ceremony

Diane Kewley Port,
*President, Acoustical Society of America*

## Annual Membership Meeting

### Presentation of Certificates to New Fellows

#### Elected in December 2019

Jonas Braasch – For interdisciplinary contributions to musical acoustics and psychoacoustics of spatial audio technology

Cynthia G. Clopper – For contributions to the acoustics and perception of dialect variation

Stanislav Emelianov – For contributions in photoacoustic, molecular, and ultrasound elasticity imaging

Michael R. Haberman – For contributions to acoustic metamaterials and heterogeneous acoustic media

Joel Mobley – For contributions to the ultrasonic characterization of dispersive systems

Rolf Mueller – For advancing knowledge of the acoustics underlying echolocation in bats

Linda Polka – For contributions to native and non-native speech perception in infants and adults

Kuangcheng Wu – For contributions to the computational evaluation of acoustic radiation from submerged structures

Xiaoming Zhang – For contributions to ultrasound surface wave elastography in biomedical imaging

#### Elected in May 2020

Jee Woong Choi – For contributions to the field of acoustic reflection and scattering from the seabed and geoacoustic inversion

Alexander L. Francis – For contributions to understanding of cognitive-linguistic processes in speech perception, and mentorship of future scientists

Efren Fernandez-Grande – For contributions to acoustic signal processing for holography and sound field reconstruction

Bozena Kostek – For contributions to musical acoustics, artificial intelligence, and education

Siu-Kit Lau – For contributions to sustainable design in building acoustics, and education

Andrew C. Morrison – For service to the Society and acoustics education outreach

Eric L. Reuter – For contributions to architectural acoustics education and service to the Society

Kainam Wong – For contributions to signal processing of acoustic vector sensor arrays

**Introduction of Award Recipients and Presentation of Awards**

David T. Blackstock Student Council Mentor Award to Patrice S. Beddor

Rossing Prize in Acoustics Education to Daniel Butko

Walter Munk Award of The Oceanography Society to Larry Mayer

R. Bruce Lindsay Award to Julien Bonnel

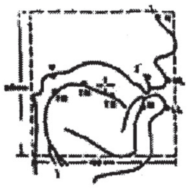
Silver Medal in Noise to Scott D. Sommerfeldt

Wallace Clement Sabine Medal to Gary W. Siebein

Gold Medal to Judy R. Dubno

Vice President's Gavel to Peggy B. Nelson

President's Tuning Fork to Victor W. Sparrow



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PART ONE - RESPIRATION, PHONATION AND AERODYNAMICS
1. The whole body plethysmograph in speech research. John J. Ohala
2. Aerodynamic end respiratory kinematic measures during speech. Elaine T. Stathopoulos
3. Physiologically based models of phonation. Ingo R. Titze
4. Use of the electroglottograph in the laboratory and clinic. James J. Mahshie
5. Endoscopy, stroboscopy, and transillumination in speech research. Anders Lofqvist, Kiyoshi Oshima

PART TWO - INDIRECT ARTICULATORY MEASUREMENTS
6. Magnetic resonance imaging (MRI) in speech research. Carol Gracco, Mark Tiede
7. Imaging the tongue with ultrasound. Maureen Stone
8. Estimating articulatory movement from acoustic data. Kenneth N. Stevens
9. Electromyography in speech research, Kiyoshi Oshima. Katherine S. Harris, Fredericka Bell-Berti

PART THREE - DIRECT ARTICULATORY MEASUREMENTS
10. The rise and fall of the soft palate: The Velotrace. Fredericka Bell-Berti, Rena A. Krakow, Dorothy Ross, Satoshi Horiguchi
11. Dynamic electropalatography. William J. Hardcastle, Fiona Gibbon
12. Measuring articulatory movements with an electromagnetic midsagittal articulometer (EMMA) system. Joseph S. Perkell, Mario A. Svirsky, Melanie L. Matthies, Joyce Manzella
13. Optoelectronic measurement of orofacial motions during speech production. Eric Vatikiotis-Bateton, Kevin Munhall, David Ostry

Each demonstration displays the instrument and how it is used; what the data look like; how data are analyzed and their applications for speech pathology, linguistics and speech processing. Anyone at any level interested in speech production and speech physiology will find these demonstrations useful. Price: $52.00

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# R. BRUCE LINDSAY AWARD

ASA
ACOUSTICAL SOCIETY
OF AMERICA

## Julien Bonnel

## 2020

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 |
| Yun Jing | 2018 |
| Adam Maxwell | 2019 |

## ENCOMIUM FOR JULIEN BONNEL

*. . . for development of physics-based signal processing methods for geoacoustic inversion and passive acoustic monitoring*

## ACOUSTICS VIRTUALLY EVERYWHERE • 9 DECEMBER 2020

Julien Bonnel grew up in Paris, spending his first 20 years in the City of Light in France. After completing his high school *baccalaureat* he enrolled in the '*classe preparatoire*', a rigorous two-year training program in the French education system that prepares and ranks highly qualified students for advanced degrees. Upon selection from this program he moved to Grenoble, near the French Alps, in 2004, to continue his studies in electrical engineering at the *Ecole Nationale Supérieure d'Ingénieurs Electriciens*. The choice of this school was likely influenced by its easy access to the ski slopes where he took up snowboarding. Over the next six years in Grenoble, he completed a Masters degree in 2007 and a Ph.D. in 2010 supervised by Jerome Mars and Barbara Nicolas at the GIPSA-Lab of the *Institut Polytechnique de Grenoble*. It was in Professor Mars' laboratory that Julien developed his interest in underwater acoustics. The French engineering university, *ENSTA Bretange*, in the small Atlantic coastal city of Brest was quick to take advantage of the opportunity to hire Julien when he graduated, and he spent the next 7 years there as head of the passive acoustics group. Throughout this time as a student and young researcher, Julien was a frequent participant at meetings of the Acoustical Society of America, and the lure of moving across the Atlantic strengthened with each visit. An offer of an Associate Scientist position from the Woods Hole Oceanographic Institution convinced him to make the move. Julien and his wife, Flora, moved to Falmouth, MA in 2017 where they live now with their 2-year old son, Sacha.

Julien Bonnel's motivating factor in research is the search for robust means of characterizing complex ocean acoustic propagation phenomena. His research is based on a synergy between signal processing and the physics of sound propagation that enables development of data processing techniques that are readily adapted to different applications in underwater acoustics. He introduced in a series of papers since 2010 an innovative approach to signal processing of broadband acoustic data-time and frequency warping. In practice, warping is a non-linear signal resampling method that compensates for dispersion in the propagating modes. The modes are transformed into a new signal space in which they appear as single tones that can be easily filtered. Julien's insightful understanding of the time-frequency properties of a broadband signal has enabled innovative use of modes for applications in geoacoustic inversion in shallow water environments, and passive detection and localization of marine mammals, two very different but very active fields of research in ocean acoustics. Julien introduced the use of warping in each of these fields and has since led the way in exploring ways to adapt the technique to new applications.

Warping made an immediate impact in research for estimating geoacoustic model parameters of the ocean bottom. With the time-frequency information in the extracted modes, it is possible to use single hydrophones instead of hydrophone arrays to estimate the geoacoustic model parameters and localize sound sources. This alone is a fundamental advance in the state of the art in geoacoustic inversion research. Previously, the conventional wisdom was that multi-element arrays were needed to resolve the modes. In addition, warping is insensitive to detailed knowledge of the ocean environment. This enables resolution of high quality observables from the acoustic signal in ocean environments that are often complicated by temporal and spatially varying features such as internal waves.

Julien recognized at an early stage in his career that signal warping would be useful in passive acoustic monitoring of marine mammals that emit low-frequency (<1 kHz) broadband sounds. The warping-based localization techniques applied to whale vocalizations enable estimation of the animal's depth in the water. This provides important clues to identify the type of whale–different whales vocalize at different depths. Over the last decade, interest from marine bioacousticians has increased year by year, and Julien has reached out to initiate interdisciplinary collaborations. To feature just a few, warping has been employed to study baleen whales in Arctic waters - to determine a northern expansion of fin-whale ranges, to track bowhead whales, and to study how sounds from a highly endangered whale species, the North Pacific Right Whale (NPRW), can be distinguished from those from more common whales. All these studies have used warping to reveal acoustic modal multipath in signals that hadn't been exploited (or even noticed) in the original recordings. Given the large numbers of shallow-water single-hydrophone recordings that exist in the bioacoustics world, it is clear that Julien's work will continue to have an insightful impact on future developments in that field.

Julien's warm sense of humor and his desire to share his ideas and insights promote collaborations in research with established researchers and young scientists that are highly productive and a great amount of fun. He is a prolific contributor to the Journal of the Acoustical Society of America, with 23 peer-reviewed papers, and 38 oral presentations (six of them invited talks at special sessions) that he has authored or co-authored with his students and collaborators. Two of his students have received best student paper awards. The Institute of Acoustics in the United Kingdom has recently recognized Julien's interdisciplinary impact in awarding him the prestigious A.B. Wood Medal in 2019.

I have no doubt that I speak for his many research colleagues and friends in offering sincere congratulations to Julien as this year's recipient of the R.B. Lindsay Award of the Acoustical Society of America. It is a well-deserved honor.

N. ROSS CHAPMAN

# ACOUSTICAL SOCIETY OF AMERICA

## Silver Medal in

## Noise


## Scott D. Sommerfeldt

## 2020

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

| | | | |
|---|---|---|---|
| Harvey H. Hubbard | 1978 | Kenneth M. Eldred | 1994 |
| Henning E. von Gierke | 1981 | Larry H. Royster | 1999 |
| William W. Lang | 1984 | Louis C. Sutherland | 2002 |
| Tony F. W. Embleton | 1986 | Alan M. Marsh | 2006 |
| William J. Galloway | 1988 | Michael R. Stinson | 2009 |
| George C. Mailing, Jr. | 1992 | Keith Attenborough | 2012 |

## CITATION FOR SCOTT D. SOMMERFELDT

*. . . for contributions to active noise and structural acoustic control.*

## ACOUSTICS VIRTUALLY EVERYWHERE • 9 DECEMBER 2020

Scott Sommerfeldt graduated Summa Cum Laude from Brigham Young University (BYU) in Provo, Utah in 1983, with a Bachelor's degree in Music Education and emphasis in clarinet performance. He then saw the light (heard the sound?) and switched over to the science of acoustics as a graduate student, earning a M.S. degree in Physics from BYU in 1986 working with Bill Strong on the acoustics of the clarinet. Scott then moved to State College, Pennsylvania as a doctoral student and earned his Ph.D. in Acoustics from the Pennsylvania State University in 1989 working with Jiri Tichy on adaptive control of vibration. Upon graduation, he accepted a position as a Research Associate and Assistant Professor at Penn State. In 1995, he saw an opportunity to revitalize the acoustics research group at BYU and accepted a faculty position in the Department of Physics and Astronomy. His dedication has resulted in several additional hires and the significant expansion of research and education activities. Scott also has a talent for administration. He served as Department Chair from 2003 to 2007 and then Dean of BYU's College of Physical and Mathematical Sciences from 2007 to 2017.

Scott is truly one of the pioneers of active noise and vibration control. Early on, he solved one of the difficult problems that had limited practical implementation – a unique method for maintaining control in time-varying environments, making adaptive control solutions much more robust. On the topic of active control, he has authored 35 peer reviewed journal articles, over half in the Journal of the Acoustical Society of America (JASA). Scott has also pioneered the use of new metrics to provide active control, incorporating loudness calculations, eigenvalue equalization, sound quality metrics, and energy-based acoustic quantities into control strategies and controller design. Many of his most-cited papers on active control of noise and vibration include this energy-based acoustics approach because it requires far fewer sensors than other approaches, sometimes only one. This effort has spawned entirely new research areas including adaptive equalization of sound fields, sound power measurements in nonideal environments, global active control of enclosures using energy density, new approaches to near-field acoustical holography, energy-density sensor design, and looking for parallel approaches in structural vibration and radiation. Scott's approach to problems—to understand the essential physics and then use that information to guide the design of solutions—has allowed him to successfully reduce the noise from aircraft engines, desktop and laptop cooling fans, noise in tractor cabs, and more.

For more than 30 years, Scott has been creating a legacy of outstanding research while also training an exceptional generation of professionals in acoustics through his mentorship. Especially noteworthy are his pioneering work in active control of noise and vibration and novel investigation of energy-based quantities in both the structural and acoustic domains. Scott has produced 66 peer reviewed journal publications and an additional 68 conference proceedings, multiple book chapters, and holds six patents. His 66 journal publications have included 51 student authors, often with them as first author, demonstrating his training of future acousticians. His mentoring of 50 students as their principal advisor has ranged from undergraduates to post-doctoral scholars. Additionally he has published with another 20 professionals, demonstrating his extensive network of collaboration. Scott's research track record is recognized by the impressive list of industry sponsors of his work: BBN, GE, Caterpillar, Intel, STI Technologies, etc., in addition to government sponsors: NSF, NSWC, and NASA.

Scott has been extensively involved in the Acoustical Society's technical activities and leadership. He recently served the ASA as Vice President in 2018, as a Member of the Executive Council, Chair of the Structural Acoustics and Vibration Technical Committee, and Chair of the Noise Technical Committee. Scott is a Fellow, has organized numerous special sessions at ASA meetings, and has served as an Associate Editor for JASA Express Letters. He was also the General Chair and General Co-Chair for the ASA meetings in Salt Lake City in 2007 and 2016, respectively. Scott has served on several additional committees and has represented ASA as liaison on education and physics resources policy committees of the American Institute of Physics (AIP). In addition to his extensive service within ASA, Scott has served as a member of the AIP Governing Board, as a member of the Institute of Noise Control Engineering (INCE) Board of Directors, as an Associate Editor for Noise Control Engineering Journal, Chair of the INCE Active Control Technical Group, General Chair for Active 99 (INCE), and Editor of conference proceedings.

One of Scott's notable qualities is his ever-present sense of humor. It has served him well as he has raised five children with his wife, Lisa, and balanced significant administrative, professional, and ecclesiastical responsibilities while also being a professor. He is skilled in telling "Dad jokes," especially when he is chairing large gatherings. He loves to teach students the "magimatics" of acoustics. He also likes to refer to the "finite elephant method." From Penn State to BYU, he has impressed upon hundreds of acoustics students the need to memorize Euler's equation just in case they're faced with a life-threatening situation where reciting it will save them from certain death. (In case you've forgotten, the linear Euler's equation is

$$\rho_0 \frac{\partial \vec{u}}{\partial t} = -\vec{\nabla} p$$

BRIAN E. ANDERSON
KENT L. GEE
TIMOTHY W. LEISHMAN

# WALLACE CLEMENT SABINE AWARD OF THE ACOUSTICAL SOCIETY OF AMERICA


## Gary W. Siebein

## 2020

The Wallace Clement Sabine Award is presented to an individual of any nationality who has furthered the knowledge of architectural acoustics, as evidenced by contributions to professional journals and periodicals or by other accomplishments in the field of architectural acoustics.

PREVIOUS RECIPIENTS

| | |
|---|---|
| Vern O. Knudsen | 1957 |
| Floyd R. Watson | 1959 |
| Leo L. Beranek | 1961 |
| Erwin Meyer | 1964 |
| Hale J. Sabine | 1968 |
| Lothar W. Cremer | 1974 |
| Cyril M. Harris | 1979 |
| Thomas D. Northwood | 1982 |
| Richard V. Waterhouse | 1990 |
| A. Harold Marshall | 1995 |
| Russell Johnson | 1997 |
| Alfred C. C. Warnock | 2002 |
| William J. Cavanaugh | 2006 |
| John S. Bradley | 2008 |
| J. Christopher Jaffe | 2011 |
| Ning Xiang | 2014 |
| David Griesinger | 2017 |
| Michael Vorländer | 2018 |

## SILVER MEDAL IN ARCHITECTURAL ACOUSTICS

The Silver Medal is presented to individuals, without age limitation, for contributions to the advancement of science, engineering, or human welfare through the application of acoustic principles, or through research accomplishment in acoustics.

PREVIOUS RECIPIENT

Theodore J. Schultz 1976

# ENCOMIUM FOR GARY WALTER SIEBEIN

*. . . for internationally recognized original and enduring contributions to the measurement, education, and practice of architectural acoustics.*

## ACOSTICS VIRTUALLY EVERYWHERE • 9 DECEMBER 2020

Gary Siebein's contributions to the field of architectural acoustics have spanned over four decades advancing, disseminating, and implementing architectural acoustics knowledge as an educator, researcher, and practitioner. He has contributed core room acoustics measurement techniques funded by the National Science Foundation (NSF) and has educated a generation of architect acousticians who are now in practice, teaching in the academy, and holding senior levels in academic administration in the US and abroad. His scholarly contributions include significant articles, papers, seminars, and book chapters and the education of thousands of undergraduate, graduate students and dozens of Master and Doctoral students specializing in acoustics. Professor Siebein has also implemented both precedent and emerging acoustical knowledge as the principal of the nationally recognized firm Siebein Acoustic, a consulting practice with projects that span from large-scale community soundscapes to detailed noise control solutions in highly technical spaces. His continuing service to the ASA and profession extends to collaboration on the refinement and development of national and international acoustics standards. As an architect acoustician, Gary's commitment and achievements in advancing acoustics through practice, research and education are outstanding and inspiring.

Gary was born in Brooklyn, New York to Walter and Peggy Siebein and lived in New York and Connecticut during his childhood. After completing his Bachelor of Science in the Building Sciences and a professional Bachelor of Architecture degree cum laude at Rensselaer Polytechnic Institute (RPI), Gary worked in architecture firms during the day as an Intern Architect and in the evenings at a local theater. He was initiated into I.A.T.S.E., the theatrical stage employee union and worked his way through college and his architectural internship. It was at the American Shakespeare Theater in Stratford, Connecticut that he was introduced to architectural acoustics while working as a stage hand. This theater was renowned for having a great acoustic environment – but why? This question intrigued Gary and he set off to learn as much as he could about acoustics.

Gary's quest for acoustics knowledge led him to Professor Harry Rodman at RPI and subsequently 1,100 miles south to the University of Florida (UF) to study with Bertram (Bert) Y. Kinzey Jr.. Bert was teaching courses in Architectural Acoustics at UF and he and Gary's relationship flourished leading to a lifelong friendship, mentorship, and partnership. Gary received his Master's degree and concentrated on research involving how the impulse response could be a measurement and diagnostic tool in both buildings and scale models and how the relationship between the two scales could allow acoustics to be accurately studied at smaller scales. His research was highlighted by being awarded a Schultz Grant by the Robert Bradford Newman Award Fund for a video demonstrating the deconstruction of an impulse response as tool to measure and evaluate sound reflecting from room surfaces as well as research awards from noted architectural journals. As Gary completed his Master's degree, Bert was preparing to retire and worked with Gary to transition into leading the Environmental Technology program and eventually the Architectural Acoustics graduate study program as an Assistant Professor at the University of Florida in 1981.

As a new faculty member, Gary's teaching and research agenda formed around natural, passive, and then lastly active (energy consuming) responses to achieve comfort in architectural spaces. He taught architecture design studios, environmental technologies courses, graduate seminar courses, soundscape labs, and doctoral core courses in the Ph.D. program including his long-time favorite the Philosophy of Inquiry. He served as mentor, thesis, and dissertation chair or committee member for dozens of masters and doctoral students and taught thousands of undergraduate students. He initiated cutting-edge research modeling acoustical phenomenon in physical scale models of performing arts and other spaces by scaling the wavelengths of sound with newly emerging state-of-the-art, super high-speed digital sampling equipment. He and his students, designed, constructed, and tested many theaters and performance spaces to evaluate the relationships between geometry and measurements and the relationship of scale models to buildings. During the late 1980's, Gary led the Master of Science in Acoustics Program at the University of Florida until he retired in 2015. His program gained the attention of some of the most renowned and well-respected acoustical consultants at the time and received substantial grants from the NSF for collaborating on new measurement systems in architectural spaces. Gary was commissioned by these firms and agencies over the years to construct and conduct acoustical measurements in scale models as part of the design process that would compare modeled and actual measurements in world-class performing spaces including the Esplanade in Singapore and the Escondido Civic Center. In 1987 and again 1994, the body of work by Gary and his students in the acoustics program was awarded the Progressive Architecture Research Award that "recognizes risk-taking practitioners and seeks to promote progress in the field of architecture" – one of the most competitive and coveted national/international awards in the field of architecture at that time.

Gary's first project as an independent acoustical consultant was an intriguing story of an otherworldly project with Phil Hawes, the Architect for the landmark Biosphere II. Hawes had read an article in Progressive Architecture about one of Gary's research projects in acoustical modeling sponsored by NSF. Gary was asked if he would be interested in working on the Biosphere II, an enclosed ecosystem to support life on Mars. This serendipitous interaction launched Gary's career as an internationally renowned acoustical consultant, and this single high profile project expanded into over 2,200 projects in locations globally during his career as an acoustical consultant – still no work on Mars, yet.

Gary has a continued and outstanding record of service to the ASA through a continuity of engagement in meetings, publishing at the forefront of evolving issues in room acoustics, measurement, noise control, serving to improve national and international acoustical standards, publishing book chapters on acoustics, and conducting seminars across the disciplines of acoustics, soundscapes, and architecture. Beyond that, he has a deep and consistent commitment to his family, those closest to him, and the broader community. In addition to Fellowship in the ASA, Gary is a member of the College of Fellows of the American Institute of Architects who honor those they refer to as 'citizen architects' – people who have a lifelong record of contributing to their profession and communities, helping their colleagues, and giving back with knowledge, kindness, and humanity. In that regard, Gary could be considered a 'citizen acoustician'. A lifelong learner and hands-on builder, Gary is constantly involved in helping his family with his wife Rita by his side. Gary's life and achievements have emerged from a deep-rooted core faith in family togetherness.

Congratulations to Gary Siebein for this well deserved award.

MARTIN GOLD
KEELY SIEBEIN

# Gold Medal

## Judy R. Dubno

## 2020

The Gold Medal is presented in the spring to a member of the Society, without age limitation, for contributions to acoustics. The first Gold Medal was presented in 1954 on the occasion of the Society's Twenty-Fifth Anniversary Celebration and biennially until 1981. It is now an annual award.

PREVIOUS RECIPIENTS

| | |
|---|---|
| Wallace Waterfall | 1954 |
| Floyd A. Firestone | 1955 |
| Harvey Fletcher | 1957 |
| Edward C. Wente | 1959 |
| Georg von Békésy | 1961 |
| R. Bruce Lindsay | 1963 |
| Hallowell Davis | 1965 |
| Vern O. Knudsen | 1967 |
| Frederick V. Hunt | 1969 |
| Warren P. Mason | 1971 |
| Philip M. Morse | 1973 |
| Leo L. Beranek | 1975 |
| Raymond W. B. Stephens | 1977 |
| Richard H. Bolt | 1979 |
| Harry F. Olson | 1981 |
| Isadore Rudnick | 1982 |
| Martin Greenspan | 1983 |
| Robert T. Beyer | 1984 |
| Laurence Batchelder | 1985 |
| James L. Flanagan | 1986 |
| Cyril M. Harris | 1987 |
| Arthur H. Benade | 1988 |
| Richard K. Cook | 1988 |
| Lothar W. Cremer | 1989 |
| Eugen J. Skudrzyk | 1990 |
| Manfred R. Schroeder | 1991 |
| Ira J. Hirsh | 1992 |
| David T. Blackstock | 1993 |
| David M. Green | 1994 |
| Kenneth N. Stevens | 1995 |
| Ira Dyer | 1996 |
| K. Uno Ingard | 1997 |
| Floyd Dunn | 1998 |
| Henning E. von Gierke | 1999 |
| Murray Strasberg | 2000 |
| Herman Medwin | 2001 |
| Robert E. Apfel | 2002 |
| Tony F. W. Embleton | 2002 |
| Richard H. Lyon | 2003 |
| Chester M. McKinney | 2004 |
| Allan D. Pierce | 2005 |
| James E. West | 2006 |
| Katherine S. Harris | 2007 |
| Patricia K. Kuhl | 2008 |
| Thomas D. Rossing | 2009 |
| Jiri Tichy | 2010 |
| Eric E. Ungar | 2011 |
| William A. Kuperman | 2012 |
| Lawrence A. Crum | 2013 |
| Brian C. J. Moore | 2014 |
| Gerhard M. Sessler | 2015 |
| Whitlow W. L. Au | 2016 |
| William M. Hartmann | 2017 |
| William A. Yost | 2018 |
| William J. Cavanaugh | 2019 |

## ENCOMIUM FOR JUDY R. DUBNO

*. . . for contributions to understanding age-related hearing loss and for leadership in the acoustics community*

### ACOUSTICS VIRTUALLY EVERYWHERE • 9 DECEMBER 2020

Judy R. Dubno is an outstanding scientist, mentor, and leader, and her contributions to acoustics and to the Acoustical Society of America (ASA) are remarkable. She has made tremendous contributions both to auditory science and to the community of people working in the field of hearing and acoustics.

Judy Dubno was born and raised in Manhattan and attended New York City public schools. One hint of an early interest in auditory science was a science fair project, in elementary school: she designed and built a color-coded clay model of the outer, middle, and inner ear. Her interest in science was nurtured while attending the Bronx High School of Science, a very competitive, specialized high school in northern New York City, which required an hour-long subway ride. Judy's interest in hearing science also came from her mother, who had a lifelong severe hearing loss in one ear resulting from an early bout of mumps.

Judy received her Ph.D. from the City University of New York (CUNY) Graduate School and University Center. Her major advisor was Harry Levitt and she was also strongly influenced by Gerald Studebaker, James (Mac) Pickett and Irv Hochberg. Her dissertation research involved predicting consonant confusions by individuals with normal and impaired hearing from the acoustic analysis of consonants. She was heavily involved in the development, recording, and analysis of the CUNY Nonsense Syllable Test, which was among the first closed-set tests that generated consonant confusions matrices (from paper and pencil responses), and is still in use today. She assembled hundreds of consonant-vowel and vowel-consonant syllables recorded on audio tape, which were cut and spliced together and labelled with a glass marker on pieces of scotch tape. Then, the pieces of audio tape were taped to the walls all over the lab for "randomization." This is quite different from the fast and accurate digital editing available today.

Judy's doctoral research was part of a large National Institute of Health (NIH)-funded contract to develop and evaluate methods for the automated selection of hearing-aid frequency-gain responses using a wearable master hearing aid, one of the first of its kind in the analog era, and employing an early form of a multivariate adaptive testing strategy. The hearing-aid settings were changed by hand using tiny modules inside the master hearing aid case, which she removed and inserted using tweezers during the experiment to adjust the listening conditions.

After receiving her Ph.D., Judy moved to Los Angeles to take an NIH-funded postdoctoral fellowship position at the University of California Los Angeles (UCLA) School of Medicine in the Division of Head and Neck Surgery, where Don Dirks and Don Morgan were significant mentors and role models. Los Angeles is where she met her husband, John, in 1985. Judy joined the faculty at UCLA, taught otolaryngology residents, and conducted research until 1991, when she moved to the Medical University of South Carolina (MUSC) in Charleston, where she remains today.

Judy's research, especially, after she joined the research group at MUSC, focused on the effects of hearing loss and age on the perception of speech and other sounds. The detailed examination of confusion matrices conducted as part of her dissertation research revealed much more about the difficulties faced by hearing-impaired people than overall error scores. This approach paved the way for many others, and the analysis of patterns of speech-sound confusions is now commonplace. Judy also led a series of seminal studies examining the specific types of acoustic cues and listening situations that lead to problems for hearing-impaired and older people. For example, she examined the role of the spatial separation of target and background sounds and the ability to benefit from amplitude fluctuations in background sounds. In her work, Judy has always been careful to try to separate the effects of age and hearing loss, whereas these effects have often been confounded by other researchers. A particularly important contribution has been Judy's leadership of a 30-year longitudinal study of age-related hearing loss and an exceptional team of research colleagues, and its work on the classification of audiograms and the use of this system to assess the underlying nature of hearing disorders. In particular, her work has been well-described in the many publications she has authored and co-authored in highly respected auditory journals. This research has been highly influential and we anticipate that its importance will increase over the coming years.

Judy has mentored a very large number of undergraduate and graduate students, post-doctoral fellows, medical-school residents, and junior faculty. Her teaching and mentoring style is nothing short of outstanding, and was recognized by her university with a mentoring award. All you have to do is talk to one of her former students or post-docs to understand what an exceptional person she is. Among those she has mentored are many who are now well-established in auditory research and teaching positions.

The excellence of Judy's research has been recognized by the many grants that she has received from NIH (continuously since 1981) and other bodies and by numerous awards. She has been elected as a Fellow of the ASA, the American Speech-Language-Hearing Association (ASHA), and the International Collegium of Rehabilitative Audiology. She received the Editor's Award from the *Journal of Speech and Hearing Research* in 1996; the Editor's Award from the journal *Ear and Hearing* in 2009; the James Jerger Career Award for Research in Audiology in 2011; the Carhart Memorial Lectureship of the American Auditory Society in 2012; Honors of the Association from ASHA, its highest award, in 2019; and the Governor's Award for Excellence in Science from the South Carolina Governor, in 2018.

We turn now to a consideration of Judy's contributions to the acoustics community. While many people can be considered "good citizens" in this respect, Judy's contributions can be regarded as exceptional, and certainly way above what most people do. She has been an editor or guest editor of six journals, she has served as the President of the Association for Research in Otolaryngology (ARO), and has served on many committees of the ARO. She has served as an officer and committee member in other scientific societies including the American Otological Society and the American Speech-Language-Hearing Association, served on four boards/committees for the National Academies of Sciences, Engineering, and Medicine, and was elected to the Collegium Oto-Rhino-Laryngologicum Amicitiae Sacrum and the International Collegium of Rehabilitative Audiology. She has also served as ASA's member society Director on the Board of Directors of the American Institute of Physics (of which ASA is a member society) and is currently its Corporate Secretary.

More recently, Judy has been participating in research and public service to support improved accessibility and affordability of hearing health care, especially hearing aids, and to enact the necessary changes in federal policies and legislation. This began in 2009 with a partnership with Amy Donahue (then at National Institute on Deafness and Other Communication Disorders) and Lucille Beck at the Veterans Administration. Judy served on a National Academies committee on this topic and is now a member of the Lancet Commission that is addressing the global burden of hearing loss.

Finally, we turn to Judy's huge contributions to the ASA. She has served on 22 committees of the Society. She was co-chair of the Vision 2020 Retreat (2011-2012), co-organizer of the ASA School in 2012, 2014, 2016, and 2018, and co-chair of ASA's first Strategic Leadership for the Future conference (2014-2015). She served as Vice President of the ASA, 2010-2011, and President, 2014-2015. She is currently the Society's Treasurer, a post carrying great responsibility that she has been fulfilling with considerable skill and dedication. In all of her service to the ASA, she has been efficient, thorough, effective, knowledgeable, empathetic, tactful, and friendly. All of those who have worked with her will attest to the fine qualities of her leadership.

Overall, the evidence is very clear that Judy Dubno amply merits the award of the Gold Medal of the ASA for her contributions to research, education, the auditory community and to the ASA itself. We congratulate her most warmly on the award of the Gold Medal of the Acoustical Society of America.

BRIAN C. J. MOORE
WILLIAM A. YOST
PEGGY B. NELSON

# Session 4aAAa

## Architectural Acoustics: Session in Honor of William J. Cavanaugh I

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

**Chair's Introduction—9:30**

### *Invited Papers*

**9:35**

**4aAAa1. In rememberance of William J. Cavanaugh.** Gregory C. Tocci (Cavanaugh Tocci, 327F Boston Post Rd., Sudbury, MA 01776, gtocci@cavtocci.com)

William J. "Bill" Cavanaugh has left us a legacy of contributions to the practice of architectural acoustics and to the Acoustical Society of America. With a B. Arch. from MIT in 1951, and following active duty as a 1st Lieutenant in the U.S. Army Corps of Engineers, Bill embarked on a career in architecture at Polaroid Corporation. While walking through Harvard Square, by chance he met his former professor, Robert B. Newman who taught architectural acoustics at MIT. Bob, always on the lookout for bright staff, invited Bill to visit the newly formed firm of Bolt, Beranek and Newman (BBN). This was a turning point that brought Bill to the top of his new profession of consulting in architectural acoustics. Bill played a major part in bring acoustics, then often regarded as a part of physics, into day-to-day architectural design. This presentation will outline Bill's professional path, which very much follows the path of the growth of architectural acoustics.

**9:55**

**4aAAa2. Bill Cavanaugh—His BBN years and his influence on architectural acoustics education.** Carl Rosenberg (Acentech, 33 Moulton St., Cambridge, MA 02138, crosenberg@ACENTECH.com) and K. Anthony Hoover (McKay Conant Hoover, Westlake Village, CA)

Bill Cavanaugh was vitally important to the architectural acoustics consulting firm of Bolt Beranek and Newman (BBN) in its early years, during his tenure (1954–1970). He managed its architectural acoustics department, and he hired and mentored many of the best known consultants from that era. He was also enthusiastic and effective in promoting acoustics classes at architectural schools, in the spirit of his mentor Robert Newman, as well as numerous educational outreaches including the celebrated Laymon Miller short courses. This presentation will offer some insight and anecdotes about the early BBN years, with implications for current and future acoustical consulting. Additionally, an overview will be offered of Bill's influence on architectural acoustics education, including the various courses at colleges and universities; the short courses and seminars for architectural firms, government, industry, professional societies, and more; and the origin and continuing good work of the Newman Student Award Fund including its Student Medals, Schultz Grants, Newsletters, and the series of concerts.



**10:15**

**4aAAa3. Speech privacy, yesterday and where we are today.** Kenneth W. Good (Armstrong World Industries, Inc., 2500 Columbia Ave., Lancaster, PA 17601, kwgoodjr@armstrong.com)

Speech privacy is one of the pillars contributing to Bill Cavanaugh's outstanding legacy. His early research in the 1950 and 1960s set a trajectory that continues to this day. This paper will discuss where we are today while honoring the roots of the topic and include an update from the TCAA Subcommittee on Speech Privacy in the Built Environment.

**10:35**

**4aAAa4. Architectural acoustics—From art to science, a tribute to Mr. William J. Cavanaugh.** Viken Koukounian (K.R. Moeller Assoc. Ltd., 3-1050 Pachino Court, Burlington, ON L7L 6B9, Canada, viken@logison.com)

The influence of Mr. William J. Cavanaugh's works, which span six decades, can be similarly summarized by the impact of his most recognized and pivotal publication—*Speech Privacy in Buildings*. The significance of this paper is appreciated in review of its place in history and following an understanding of the discoveries it permitted. Just as Mr. Cavanaugh balanced complex subjective and objective measures between various topics of study (acoustics, electronics, psychology, and physiology) in his novel work, he has empowered us to create and ensure a more favorable balance between art and science in architectural acoustics.

## Session 4aAAb

# Architectural Acoustics: Session in Honor of William J. Cavanaugh II

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

**Chair's Introduction—11:15**

## *Invited Papers*

**11:20**

**4aAAb1. Teaching architectural acoustics, as inspired by Bill Cavanaugh.** K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com) and Eric Reuter (Reuter Assoc., Portsmouth, NH)

Bill Cavanagh led by example in teaching architectural acoustics. This presentation will look back at the 1-1/2 credit classes in acoustics that he taught at various colleges, and which inspired and influenced many subsequent acoustics classes and programs. Also, one outgrowth from these earlier classes will be discussed, the acoustics class at Berklee College of Music—its origin, development, and outlook—which over 33 years has provided a three credit class to over 7000 students who are deeply immersed in audio and music, and now have a solid foundation in acoustics.

**11:40**

**4aAAb2. Architectural acoustics, co-edited by Bill Cavanaugh.** Jin Yong Jeon (Dept. of Architectural Eng., Hanyang Univ., 222 Wangsimni-ro, Seongdong-gu, Seoul 04763, South Korea, jyjeon@hanyang.ac.kr)

Architectural acoustics, in addition to heat, light, and air, is an important component of architectural environment planning. Several monographs have been published in recent years. Among these, the best and most widely known is "Architectural Acoustics," co-edited by William Cavanaugh (acoustician) and Joseph Wilkes (architect). It consists of a collection of chapters on various aspects of architectural acoustics, written by Bill and other experts. Bill's main contribution, by way of this book, "Architectural Acoustics," to the field is the practical explanation of applicable architectural acoustic knowledge. In the second edition (copyrighted 1999) especially, various case studies involving a number of acoustic consultants in the second half of each chapter allow students to acquire sound designs through practical application and scientific verification. When this textbook was introduced and taught to masters and doctoral students in the field of acoustics, they learned that architectural acoustics was an extremely practical field of study. Most graduates who studied this book continued their studies and earned their Ph.D.s and are now teaching in universities or working as researchers in research institutes.

**12:00**

**4aAAb3. Outdoor concert monitoring, community noise, and the Cavanaugh Criterion.** K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com) and Bennett M. Brooks (Brooks Acoust. Corp., Pompano Beach, FL)

Sound from outdoor concerts and other sources can be challenging to measure, especially in the midst of ambient noises and environmental effects. It can be even more challenging to evaluate the data toward resolution of complaints and satisfaction for neighbors. In the mid 1980s through the 1990s, Bill Cavanaugh was deeply involved in important projects involving measurement and especially evaluation of outdoor concert sound levels, and he became a recognized leader in the field, consulting on amphitheater and outdoor venue sound and addressing community relations throughout the country. This presentation will review one of the most significant projects, then called Great Woods Center for the Performing Arts, now the Xfinity Center, in Mansfield, MA, an outdoor amphitheater with total 16 000 seats (now nearly 20 000), in which the Cavanaugh Criterion took shape, more in response to the many touring rock shows (and for which one of the first "green-yellow-red lights" was developed for the sound mixers' easy visual reference and better sound level control) than as the summer home of The Pittsburgh Symphony Orchestra. Recent application of the Cavanaugh Criterion for concert level evaluation of loud, powerful sound systems will also be reviewed, along with implications for application to a much wider range of possible outdoor and even indoor noise intrusions.

12:20

**4aAAb4. Bill Cavanaugh: Contributions to the understanding of room acoustics in performance venues through the Concert Hall Research Group.** Robin Glosemeyer Petrone (Threshold Acoust., 141 W Jackson Blvd, Ste. 2080, Chicago, IL 60604, robin@thresholdacoustics.com)

A remarkable role model for acoustics consultants at any point in their career, Bill Cavanaugh was kind, humble, and always willing to share his successes and failures in hopes that we would all be better acousticians. In 1991, several acoustic consulting firms, including Cavanaugh Tocci Associates, came together to form the Concert Hall Research Group. They pooled experience and funds to support research and measurements in the interest of advancing the understanding of acoustics in concert halls. When the original work of collecting measurements from several halls was completed, Bill championed the idea of creating a Summer Institute to continue the advancement of knowledge. The Institute brought together acoustic consultants in performing arts, faculty and researchers in architectural acoustics, and college students studying or interested in architectural acoustic to foster conversations and honest sharing of knowledge in an informal and intimate setting.

THURSDAY MORNING, 10 DECEMBER 2020 — 9:30 A.M. TO 11:00 A.M.

# Session 4aAB

## Animal Bioacoustics: Effects of Anthropogenic Noise on Animal Behavior

Bethany Holtz, Chair

*Environmental Studies, Gettysburg College, 183 Comanche Trail, Medford Lakes, NJ 08055*

**Chair's Introduction—9:30**

### *Contributed Papers*

9:35

**4aAB1. Effects of vessels and their sounds on the foraging behavior of endangered killer whales (*Orcinus orca*).** Marla M. Holt (Conservation Biology Div., NOAA/NMFS Northwest Fisheries Sci. Ctr., 2725 Montlake Blvd East, Seattle, WA 98112, marla.holt@noaa.gov), Jennifer Tennessen (Lynker Technologies and NOAA Northwest Fisheries Sci. Ctr., Seattle, WA), M. Bradley M. Hanson, Candice Emmons (Conservation Biology Div., NOAA/NMFS Northwest Fisheries Sci. Ctr., Seattle, WA), Deborah Giles (Univ. of Washington Friday Harbor Labs., Davis, CA), and Jeffrey Hogan (Cascadia Res. Collective, Olympia, WA)

Anthropogenic activities that have negative consequences on foraging outcomes warrant special concern in endangered species. Prey availability and vessel disturbance are identified risk factors of endangered Southern Resident killer whales (SRKW) as vessels and associated sounds can mask echolocation signals used for foraging and/or disrupt foraging behavior with implications for energy acquisition. To investigate vessel effects on foraging behavior, we utilized multi-sensor, digital acoustic recording tags (Dtags) that were suction cup-attached to SRKW. We tested a number of explanatory variables related to nearby vessels and associated sounds on the probability of prey capture and prey capture dive parameters. Tested variables included echosounder presence/absence, vessel counts, vessel distance, and vessel speed, and several competing received noise level measurements (absence of flow noise). Although noise level was not a significant explanatory variable on the probability of prey capture, flow noise limited available sample size. Furthermore, the probability of prey capture decreased as vessel speed increased which correlates with vessel noise. We also found a significant effect of echosounder presence on the duration and rate of descent of prey capture dives, indicating further consequences on foraging behavior. These results inform conservation and management efforts to preserve SRKW foraging opportunities and mitigate vessel disturbance.

9:55

**4aAB2. Beaked whale foraging behavior during 12 kHz multibeam mapping survey off southern California coast.** Hilary S. Kates Varghese (Ctr. of Coastal and Ocean Mapping, Univ. of New Hampshire, 24 Colovos Rd., Durham, Durham, NH 03824, hkatesvarghese@ccom.unh.edu), Jennifer Miksis-Olds (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, Durham, NH), Nancy DiMarzio (Ranges, Eng. and Anal. Dept., Naval Undersea Warfare Ctr., Newport, RI), Kim Lowell (Ctr. of Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH), Ernst Linder (Dept. of Mathematics and Statistics, Univ. of New Hampshire, Durham, NH), David Moretti (Ranges, Eng. and Anal. Dept., Naval Undersea Warfare Ctr. (retired), Newport, RI), and Larry A. Mayer (Ctr. of Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH)

To create appropriate marine mammal protection regulations, a better understanding of potential impacts of anthropogenic noise on marine mammals is needed. To contribute to the growing body of knowledge on this subject, echolocation clicks of foraging Cuvier's beaked whales were detected on the Southern California Antisubmarine Warfare Range (SOAR) hydrophones during 12 kHz multibeam echosounder (MBES) ocean mapping surveys in 2017 and 2019, and assembled into foraging events called group vocal periods (GVPs). Four GVP characteristics were analyzed *Before, During,* and *After* the MBES surveys to assess differences in foraging behavior with respect to the mapping activity. The number of GVP per

hour increased *During* and *After* MBES surveys compared with *Before*. There were no other differences between non-MBES and MBES periods for the other characteristics: number of clicks per GVP, GVP duration, and click rate. These results indicate that the animals did not leave the range or stop foraging during MBES activity. [Funded by NOAA Grant No. NA15NOS4000200 through the UNH Joint Hydrographic Center.]

**10:15**

**4aAB3. Interactions between whale-watching boats and humpback whale (*Megaptera novaeangliae*) singing cycle in the Colombian North Pacific soundscape.** Maria P. Rey-Baquero (Estudios Ambientales y Rurales, Pontificia Universidad Javeriana, Bogota 111511, Colombia, rey_m@javeriana.edu.co), Laura V. Huertas-Amaya (Estudios Ambientales y Rurales, Pontificia Universidad Javeriana, Bogotá, Colombia), Kerri D. Seger (Appl. Ocean Sci., Seattle, Fairfax Station, VA), Natalia Botero-Acosta (Fundación Macuáticos Colombia, Bogotá, Colombia), Christina E. Perazio (Neural and Cognit. Plasticity Lab, State Univ. of New York Univ. at Buffalo, Evolution, Ecology, and Behavior, Buffalo, Buffalo, NY), and Andrea Luna-Acosta (Estudios Ambientales y Rurales, Pontificia Universidad Javeriana, Bogotá, Colombia)

Noise in marine ecosystems has increased significantly in the last decades. One of the most significant sources is vessel traffic. This affects animals that depend on sound to interact within their ecosystems, like interrupting communication which could lead to adaptive strategies to avoid the noise. We recorded the soundscape from 0 to 15 kHz in Morro Mico and Nuquí, Colombia, from 2018 and 2019—one of the most biodiverse locations in the world, and one that has never experienced high levels of vessel traffic and has no shipping lanes. Here, humpback whale Stock G migrates to their breeding ground. The species is well-known for its songs and social calls that facilitate biological and ecological processes. One process that could shape these vocalizations is noise generated by the artisanal fishing family boats. We explored correlations between the sound energy from the humpback whale song cycle, temporal features of the song structure, other biological acoustic signals, physical acoustic cycles, and the presence of small boats. These comparisons are important because there are no large vessels (requiring AIS) that transit closely to the study area. Therefore, we focused on soundscape cycles and acoustic responses of humpback whales from purely small vessel noise.

**10:35**

**4aAB4. Influence of environmental and anthropogenic acoustic cues in sea-finding of hatchling Leatherback *(Dermochelys coriacea)* sea turtles.** Bethany Holtz (Biology Dept., Saint Joseph's Univ., 5600 City Ave., Philadelphia, PA 19131, bethany.holtz94@gmail.com), Wendy Dow Piniak (Div. of Marine Sci. and Conservation, Marine Lab., Nicholas School of the Environment, Duke Univ., Beaufort, NC), and Kelly Stewart (The Ocean Foundation, Washington, DC)

Observed visual and geomagnetic orientation cues on sea turtle hatchlings overlook the impact auditory stimuli may have on orientation. We hypothesized that increased levels of anthropogenic sound, linked to behavioral and physiological cues found in other marine species, impact hatchlings during sea-finding. The responses of hatchling leatherbacks, *Dermochelys coriacea,* collected from the Sandy Point National Wildlife Refuge, St. Croix, were measured in the presence of aerial acoustic sounds within hatchlings' hearing range of 50 to 1600 Hz. The highest sound energy produced by beach waves occurs at frequencies 50–1000 Hz, which overlaps with the most sensitive hearing range of hatchling leatherbacks (50–400 Hz). Natural beach wave sounds, which peak at frequencies of 50–1000 Hz, may be masked by human conversations (85–650 Hz) and vehicle traffic (60–8000 Hz). In our controlled experimental design, we exposed hatchlings to surf wave sounds (70 dB *re*: 20 $\mu$Pa), human conversation (68 dB *re*: 20 $\mu$Pa), and traffic noise (70 dB *re*: 20 $\mu$Pa). We observed no phonotaxic response (wave sounds: mean angle = 152.1 deg, $p = 0.645$; human conversation: mean angle = 67.4 deg, $p = 0.554$; traffic noise: mean angle = 125.7 deg, $p = 0.887$). This lack of orientation may be due to hatchlings' inability to localize. Visual and auditory cues may also converge to effect sea finding orientation.

## Session 4aAOa

# Acoustical Oceanography, Interdisciplinary, and Underwater Acoustics: General Topics in Acoustical Oceanography V

Ryoichi Iwase, Chair
*IMG, JAMSTEC, 3173-25 Showa-machi, Kanazawa-ku, Yokohama 236-0001, Japan*

**Chair's Introduction—9:30**

## *Contributed Papers*

**9:35**

**4aAOa1. Classification of dispersive calls using a convolutional neural network.** Mark Goldwater (Woods Hole Oceanographic Inst., 1000 Olin Way MB 432, Needham, MA 02492-1200, mgoldwater@olin.edu), Julien Bonnel, and Daniel P. Zitterbart (Woods Hole Oceanographic Inst., Woods Hole, MA)

Low-frequency acoustic signals in shallow water are highly impacted by interactions with the sea surface and seabed. The acoustic field is then conveniently described by modal theory, and the received signal can be modeled by a set of modes that propagate dispersively. It is now well established that the time-frequency dispersion of normal modes, as measured with a single hydrophone, can be used to localize the source and/or estimate the propagation environment. This method has notably been used to range vocalizations from baleen whales in shallow water. However, this method requires at least two modes to be present in the recorded call. Here, we use a convolutional neural network (CNN) to detect and classify dispersed gunshots (impulse calls) from Southern right whales, using a dataset recorded in Baja de San Antonio, Argentina. The CNN outputs the confidence of an input belonging to the following three classes: at least two modes, less than two modes, and no call present. We show that the CNN can isolate multimodal dispersive gunshots from large audio data with high precision. Such signals can then be further processed to localize the source and/or characterize the environment. [Work supported by the Office of Naval Research.]

**9:55**

**4aAOa2. Acoustic propagation and reverberation in shallow water with a gassy bottom: Experiments in the Sea of Galilee and preliminary data analysis.** Andrey Lunkov (Prokhorov General Phys. Inst. of the Russian Acad. of Sci., 38 Vavilov St., Moscow 119991, Russian Federation, lunkov@kapella.gpi.ru), Boris Katsnelson (Marine Geosciences, Univ. of Haifa, Haifa, Israel), Anatoliy N. Ivakin (Univ. of Washington, Seattle, WA), and Regina Katsman (Marine Geosciences, Univ. of Haifa, Haifa, Israel)

Broadband (200—2000 Hz) pulses were transmitted by an omnidirectional source in a shallow lake (Sea of Galilee, Israel, ~40 m depth) in several experiments during last five years. Data analysis and modeling results are presented for acoustic pressure received by a vertical line array located at an anchored station in the deepest place of the lake. Received signal was comprised of a series of arrivals followed by reverberation codas. Modeling of the propagation showed that arrival peaks' magnitudes are controlled by bottom reflectivity affected mostly by averaged (horizontally) acoustic parameters of the gassy sediment. Analysis of reverberation codas can provide further information about random spatial distribution of gas bubbles in the sediment layer and, also, about bottom roughness. Possibilities of inversions for water column and gassy bottom parameters based on these and future experiments are discussed. [Work supported by RFBR, BSF, and ONR.]

**10:15**

**4aAOa3. Long-lasting acoustic resonant signal associated with the turbidity current triggered by the 2003 Tokachi-oki earthquake.** Ryoichi Iwase (JAMSTEC, 3173-25 Showa-machi, Kanazawa-ku, Yokohama 236-0001, Japan, iwaser@jamstec.go.jp) and Shun Nomura (JAMSTEC, Yokohama, Japan)

Turbidity current triggered by the 2003 Tokachi-oki earthquake of M8 is observed at a multidisciplinary submarine cabled observatory deployed in southern Kuril subduction zone offshore of Hokkaido in Japan. The cabled observatory consists of a cable-end station, three ocean bottom seismometers (OBSs) and two pressure gauges (PGs). At each cable-end station and OBS, one hydrophone, whose sampling rate is 100 Hz, is also installed. The epicenter is located at 25 km west-northwest of the cable-end station. The arrival of the turbidity current is detected with both the electro-magnetic current meter and the acoustic Doppler current profiler (ADCP) at the cable-end station about 2 h after the earthquake occurred. In the waveform of the hydrophone at the cable-end station, relatively broadband signal is recognized to continue since the arrival of the turbidity current. About eight hours later, the peak frequency changes to 1 Hz and this resonant signal continues about four hours. Meanwhile, similar signals are not observed at OBS1 which is located 5 km northeast of the cable-end station. Although the source mechanism of the resonant signal is unknown to date, this indicates that the main body of the turbidity current passes the western side of the cable-end station.

**10:35**

**4aAOa4. A convolutional neural network applied to Arctic acoustic recordings to identify soundscape components.** Malek Ibrahim (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78558, shmeek8@gmail.com), Jason D. Sagers, and Megan S. Ballard (Appl. Res. Labs.,Univ. of Texas, Austin, TX)

Underwater acoustic recordings often have contributions from anthropogenic, geophonic, and biophonic sources and deconstructing the sound field into constituent parts can be useful when characterizing the soundscape. This work focuses on the analysis of a yearlong acoustic measurement made on the Chukchi Shelf as part of the Canada Basin Acoustic Propagation Experiment (CANAPE). The data were recorded on a shallow-water array with horizontal and vertical apertures and span time periods including the open water season, marginal ice zone, and complete ice cover. A supervised convolutional neural network is trained on labeled spectra and demonstrates the ability to separate anthropogenic and biologic sources. Details of the model, labeling process, initial results, and lessons learned will be discussed in this talk. [Work sponsored by ONR.]

## Session 4aAOb

# Acoustical Oceanography: General Topics in Acoustical Oceanography VI

Tsu Wei Tan, Chair
*Physics, Naval Postgraduate School, No. 669,Junxiao Rd., Zouying District, ROC Naval Academy, Kaohsiung 81345, Taiwan*

**Chair's Introduction—11:15**

## *Contributed Papers*

**11:20**

**4aAOb1. Applying acoustic noise to probe the water column in a dynamic shallow-water environment.** Tsu Wei Tan (Marine Sci., ROC Naval Acad., Dept. of Marine Sci., ROC Naval Acad., Kaohsiung 81345, Taiwan, ttan1@nps.edu) and Oleg A. Godin (Phys., Naval Postgrad. School, Monterey, CA)

Acoustic noise interferometry is applied to retrieve empirical Green's functions (EGFs) from the ambient and shipping noise data acquired in the Shallow Water 2006 experiment on the continental shelf off New Jersey. Despite strong internal wave-induced perturbations of the sound speed in water, EGFs are found on 31 acoustic paths by cross-correlating the noise recorded on a single hydrophone with noise on the hydrophones of a linear array about 3.6 km away. Two fifteen day-long datasets are considered. Dispersion curves of three low-order normal modes at frequencies below 110 Hz are extracted from the EGFs with time-warping technique. The dispersion curves from the first dataset were previously employed to estimate the seabed properties [T. Tan *et al.*, *J. Acoust. Soc. Am.* **147**, EL453–EL459 (2020)]. Here, using this geoacoustic model, we invert the differences between the higher-frequency part of the dispersion curves obtained from the two datasets for the variation of the time-averaged sound speed profile in water between the two observation periods. Results of the passive inversion are compared to the ground truth derived from *in situ* temperature measurements. The effect of temporal variability of the water column during noise averaging time on EGF retrieval is discussed and quantified.

**11:40**

**4aAOb2. A quantitative soundscape analysis of the Canadian Arctic, 2014–2019.** William D. Halliday (Wildlife Conservation Society, 169 Titanium Way, Whitehorse, YT Y1A 0E9, Canada, whalliday@wcs.org), David R. Barclay, Emmanuelle Cook (Oceanogr., Dalhousie Univ., Halifax, NS, Canada), Jackie Dawson (Dept. of Geography, Environment and Geomatics, Univ. of Ottawa, Ottawa, ON, Canada), John Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), Casey Hilliard (Inst. for Big Data Analytics, Dalhousie Univ., Halifax, NS, Canada), Nigel Hussey (Dept. of Biological Sci., Univ. of Windsor, Windsor, ON, Canada), Joshua Jones (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), Francis Juanes (Dept. of Biology, Univ. of Victoria, Victoria, BC, Canada), Marianne Marcoux, Andrea Niemi (Freshwater Inst., Fisheries and Oceans Canada, Winnipeg, MB, Canada), Shannon Nudds (Bedford Inst. of Oceanogr., Fisheries and Oceans Canada, Dartmouth, NS, Canada), Matthew Pine (Wildlife Conservation Society Canada, Victoria, BC, Canada), Clark Richards (Bedford Inst. of Oceanogr., Fisheries and Oceans Canada, Dartmouth, NS, Canada), Kevin Scharffenberg (Freshwater Inst., Fisheries and Oceans Canada, Winnipeg, MB, Canada), Kristin Westdal (Oceans North, Vancouver, BC, Canada), and Stephen Insley (Wildlife Conservation Society Canada, Whitehorse, YT, Canada)

In the Arctic, sound levels have historically been strongly tied to sea ice and wind speed, with very little impact of anthropogenic noise. However, climate change is causing a loss of sea ice, and consequently increased ship traffic and anthropogenic underwater noise. Here, we present the first quantitative, comparative analysis of underwater sound levels across the Canadian Arctic. We analyzed 39 passive acoustic datasets collected throughout the Canadian Arctic from 2014 to 2019 to examine spatial and temporal trends in sound pressure levels (SPL), quantify environment drivers of SPL, and estimate the influence of ship traffic on SPL. Daily mean SPL in the 50–1000 Hz bandwidth ranged from 70 to 127 dB *re* 1 $\mu$Pa (median = 91 dB). SPL increased as wind speed increased, but decreased as both ice concentration and air temperature increased. The highest SPLs were in August-October, and the lowest in March–April. SPL increased as the number of ships increased. The highest mean SPLs were recorded near southeast Baffin Island, but the most ship noise was recorded near Pond Inlet (>1 ship/day in summer). This study provides an important baseline for underwater sound levels in the Canadian Arctic, and fills many geographic gaps on published underwater sound levels.

**12:00**

**4aAOb3. Statistical analysis and modeling of wind-generated ocean noise in the northeast Pacific Ocean.** Felix Schwock (Elec. and Comput. Eng., Univ. of Washington, 185 Stevens Way, Paul Allen Ctr. – Rm. AE100R, Seattle, WA 98195, fschwock@uw.edu) and Shima Abadi (Elec. and Comput. Eng., Univ. of Washington, Seattle, WA)

In the absence of any dominant ocean noise sources, such as rain or shipping events, wind-generated noise sets the baseline for the underwater acoustic environment. In this study, we have evaluated approximately 3.5 years of acoustical and meteorological data recorded at the northeast Pacific Ocean continental shelf and slope. The acoustic data are recorded continuously at a sample rate of 64 kHz at 81 m depth and 581 m depth at the continental shelf and slope, respectively. The wind speeds are provided by a surface buoy located in the vicinity of each hydrophone. The wind speed values vary between 0 and 18 m/s with a precision of 2%. Power spectral levels are calculated to characterize the wind-generated ocean noise. A linear regression is applied to the average power spectra for different wind speeds. Our results show that wind-generated noise is in good agreement with previous studies for wind speeds less than 10 m/s. However, for wind speeds greater than 10 m/s, the spectral levels decrease significantly faster than what is reported in the literature.

**12:20**

**4aAOb4. Acoustical properties of *T*-waves generated by 2011 Tohoku earthquake.** Myungkwon Ko (Hanyang Univ., Hanyang Dae Hak ro 55, Ansan 15588, South Korea, ko.myungkwon@gmail.com), Hyun-jung Ryu (DSEM, Geoje, South Korea), Sung Won Shin (Hanyang Univ., Ansan, South Korea), Sun-Cheon Park (Korea Meteorological Inst., Seoul, South Korea), and Jee Woong Choi (Hanyang Univ., Ansan, South Korea)

*T*-wave is an underwater acoustic wave generated by submarine earthquake and propagated to a long distance through the SOFAR channel.

*T*-wave can be received by a hydrophone located in the minimum sound velocity layer of the SOFAR channel, and thus can be applied to the detection of submarine earthquake. In this study, the *T*-waves received by hydrophones installed at International Monitoring System hydroacoustic station, HA11 after being generated by 2011 Tohoku earthquake was analyzed to investigate the possibility of early tsunami warning. The station HA11 is located near Wake Island, about 3100 km from the epicenter of the Tohoku earthquake. Transmission losses as a function of frequency were predicted using a range-dependent acoustic model, RAM and used to predict the source spectral density of the *T*-wave. The results presented in this talk may be applied to the tsunami early warning system. [This research was supported by the Korea Meteorological Institute (Grant No. KMI2018-02510).]

THURSDAY MORNING, 10 DECEMBER 2020 — 9:30 A.M. TO 11:00 A.M.

## Session 4aBAa

## Biomedical Acoustics and Signal Processing in Acoustics: New Developments in Lung Ultrasound I

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

Marie Muller, Cochair
*MAE, North Carolina State University, 911 Oval Drive, Engineering Building III, Raleigh, NC 27606*

**Chair's Introduction—9:30**

### *Invited Papers*

**9:35**

**4aBAa1. Introduction of lung ultrasound surface wave elastography.** Xiaoming Zhang (Mayo Clinic, 200 1st St. SW, Rochester, MN 55905, zhang.xiaoming@mayo.edu)

Ultrasonography is not widely used for lung assessment because ultrasound cannot image deep lung tissue. Many lung diseases including interstitial lung disease (ILD) are associated with changes in lung's elastic properties. ILD includes multiple serious lung disorders associated with lung fibrosis and can lead to respiratory failure and pulmonary hypertension. High-resolution CT (HRCT) is the clinical standard for diagnosing lung fibrosis, but it results in increased radiation exposure for patients. Lung fibrosis is associated with stiffened lung parenchyma. Current clinical techniques including HRCT cannot measure lung elastic properties. Most ILDs are typically distributed in the lung's peripheral and subpleural regions. We developed noninvasive lung ultrasound surface wave elastography (LUSWE) to measure lung surface wave speed safely and quickly. In this abstract, we will present some basics for LUSWE. We will also present its clinic application for assessing ILD. We studied 30 healthy controls, 16 mild ILD patients, 45 moderate ILD patients, and 27 severe ILD patients. Lung surface wave speed has good sensitivity and specificity for separating ILD patients from healthy subjects as well as for separating mild ILD patients from healthy subjects. LUSWE is useful for other lung disorders such as pulmonary edema.

**9:55**

**4aBAa2. Identifying likely malignant lymph nodes in lung cancer through endobronchial ultrasound strain elastography and multi-modality imaging in a multi-center international prospective study.** Roel L. Verhoeven (Dept. of Pulmonology/Medical Ultrasound Imaging Ctr. (Dept. of Radiology), Radboudumc, Geert Grooteplein Zuid 10, PO Box 9101 (614), Nijmegen, Gelderland 6500HB, The Netherlands, Roel.LJ.Verhoeven@radboudumc.nl), Rocco Trisolini, Fausto Leoncini (Interventional Pulmonology Unit, Dipartimento di Scienza, Mediche e Chirurgiche, Fondazione Policlinico A. Gemelli, Rome, Italy), Piero Candoli (Pulmonology Unit, Santa Croce Hospital, Fano, Italy), Michela Bezzi (Dept. of Pulmonology, Azienda Ospedaliera Universitaria di Careggi, Firenze, Italy), Alessandro Messi (Dept. of Pulmonology, Ospedale Santa Maria Bianca, Modena, Italy), Mark Krasnik (Dept. of Pulmonology, Rigshospitalet, Copenhagen, Denmark), Chris L. de Korte (Medical Ultrasound Imaging Ctr. (Dept. of Radiology), Radboudumc, Nijmegen, The Netherlands), Jouke T. Annema (Dept. of Respiratory Medicine, Amsterdam Univ. Medical Centers, Amsterdam, The Netherlands), and Erik H. van der Heijden (Dept. of Pulmonology, Radboudumc, Nijmegen, The Netherlands)

To decide upon staging and concurrent treatment options in lung cancer, systematic endosonographic evaluation of lymph nodes as adjacent to the central airways and esophagus is of paramount importance. Based on imaging findings, repeated nodal sampling by endosonography guided needle aspiration is decided upon. Ultrasound strain elastography imaging might help identify likely malignant nodes, hypothesizing lower relative lymph node strain to be correlating to a higher chance of malignancy. Assessing if strain elastography can predict individual lymph node malignancy, also when further combined with available FDG-PET and nodal sizing information.

A multicentric international prospective trial measuring patients with a lung cancer (suspicion) in standardized fashion (n = 5). Measurement outcomes are correlated to individual node follow-up outcome. A total of 525 lymph nodes (327 patients) are included. Receiver Operator Characteristic analysis of strain elastography measurements shows an area under the curve of 0.77. Specifying a mean nodal strain <115 (0–255) indicates malignancy with sensitivity 90%, specificity 43%, positive predictive value 60% and negative predictive value 82%. Combining available PET-CT and size information with strain elastography findings allows further risk stratification. Endobronchial ultrasound strain elastography helps predict lymph node malignancy in the work-up of lung cancer.

**10:15**

**4aBAa3. Novel uses of ultrasound in various lung diseases.** Thomas M. Egan (Surgery and Biomedical Eng., Univ. of North Carolina at Chapel Hill, 3040 Burnett-Womack Bldg., Chapel Hill, NC 27599-7065, thomas_egan@med.unc.edu), Kaustav Mohanty (North Carolina State Univ., Raleigh, NC), Mir Ali, John Blackwell (Surgery, Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Micah Ulrich (North Carolina State Univ., Raleigh, NC), and Marie Muller (MAE, North Carolina State Univ., Raleigh, NC)

Conventional ultrasound (CUS) has been used in the chest to visualize pleural and pericardial fluid collections, diaphragm motion, and blood flow through the heart for anatomic definition and determining pressure gradients across valves. CUS is NOT suitable for *imaging* lung parenchyma, because of ultrasound multiple scattering (USMS) from millions of air-filled alveoli providing air-tissue interfaces. We used USMS to characterize lung tissue. We measured scattered mean free path (SMFP), the average distance between two scattering events, related to the amount of air present in lung parenchyma, and backscatter frequency shift (BFS), a measure of ultrasound attenuation. Using a rat bleomycin pulmonary fibrosis model, we showed that SMFP was longer in 18 rat lungs with pulmonary fibrosis, compared to 6 normals, and correlated with severity of lung fibrosis assessed by *ex vivo* CT scan and inflation-fixed histology. SMFP was significantly longer in 6 rat lungs made edematous by ischemia-reperfusion injury with significantly lower BFS. These data support using USMS to characterize severity of pulmonary fibrosis and pulmonary edema due to heart failure, and pneumonia and congestion in humans. In seven large animal lungs, determining the *absence* of USMS allows localization of pulmonary nodules for minimally invasive surgery. USMS provides many clinical opportunities.

## *Contributed Paper*

**10:35**

**4aBAa4. Usability of large animal models for lung ultrasound training and research.** Frank Wolfram (Lung Cancer Ctr./Clinic of Thoracic and Vascular Surgery, SRH Wald-Clinic, Str. des Friedens, Gera 07548, Germany, Frank.Wolfram@SRH.de), Holger Gutsche (Lung Cancer Ctr./Clinic of Thoracic and Vascular Surgery, SRH Wald-Clinic, Gera, Germany), Conny Braun (Animal Experimentation, Univ. Hospital Jena, Jena, Germany), and Thomas G. Lesser (Lung Cancer Ctr. / Clinic of Thoracic and Vascular Surgery, SRH Wald-Clinic, Gera, Germany)

Lung ultrasound (LUS), particular under aspects of pneumonia and cardiopulmonary insufficiency, requires specialised sonographic expertise as well as knowledge of its specific features. Due to restricted access to clinical cases such as with viral pneumonia, alternatively large animal models can be used for education and research. Therefore this work summarizes approaches for inducing pathologic LUS features on large animals. These will be classified regarding presentation of A/B-Lines, White Lung (WLS), pleural irregularities, consolidations and are further discussed regrading their cardio-pulmonary stability. Most commonly ARDS inducing lung lavage, oleic acid injection and endobronchial saline administration were published. Dominantly B-lines, WLS were found in all such models. Albeit a specialized technique using endobronchial saline instillation (OLF) visualised all LUS features as typical present during viral pneumonia, such as consolidations with air/fluid bronchogram, pleural irregularities and their patchy distribution with areas of aerated lung. With regard to haemodynamic and oxygenation stability, OLF provides best haemodynamic stability and enables reusable usability of animals. Non-infective large animal models can be used for safe, reliable education of clinicians for teaching practical handling of lung ultrasound and could serve for research aspects regarding ultrasound imaging or safety.

## Session 4aBAb

# Biomedical Acoustics and Signal Processing in Acoustics: New Developments in Lung Ultrasound II

Libertario Demi, Cochair
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Marie Muller, Cochair
*MAE, North Carolina State University, 911 Oval Drive, Engineering Building III, Raleigh, NC 27606*

**Chair's Introduction—11:15**

### *Invited Papers*

**11:20**

**4aBAb1. On the necessary paradigm-shift from qualitative to quantitative lung ultrasound.** Libertario Demi (Information Eng. and Comput. Sci., Univ. of Trento, via sommarive 9, Trento, Italia 38123, Italy, libertario.demi@unitn.it)

Lung ultrasound (LUS) is spread in the medical world since the nineties. In fact, despite ultrasound imaging is not designed for the lungs, there exists a clear difference between images of healthy and diseased lung tissue. This difference relies on specific imaging artifacts, which have been studied and classified in the literature. However, this differentiation is performed by visual inspection of clinical experts. Moreover, standardization of imaging protocols is lacking. As a result, three strong limitations affect LUS: (1) only qualitative analyses are performed, which are influenced by intra and inter operator variability; (2) intra and inter ultrasound-scanner variability is not taken into account; (3) although the sensitivity is reported to be high, specificity remains low. To tackle these problems, a paradigm shift from qualitative to quantitative LUS is needed. In this talk, current efforts in this direction will be reviewed. A standardized image acquisition protocol and scoring system developed for COVID-19 patients will be presented, together with dedicated algorithms able to automatically score and segment LUS images. Data from a multicenter study involving 200 patients will be discussed. Moreover, the results from a recent study on quantitative lung ultrasound spectroscopy applied to the diagnosis of pulmonary fibrosis will be introduced.

**11:40**

**4aBAb2. Lung ultrasound and high-resolution CT-scan of the chest for COVID-19 pneumonia.** Andrea Smargiassi (Pulmonary Medicine Unit, Dept. of Medical and Surgical Sci., Fondazione Policlinico Universitario Agostino Gemelli IRCCS, Largo Gemelli, 8, Rome 00168, Italy, andrea.smargiassi@policlinicogemelli.it), Gino Soldati (Diagnostic and Interventional Ultrasound Unit, Valle del Serchio General Hospital, Castelnuovo di Garfagnana (Lucca), Italy), Tiziano Perrone (Emergency Dept., Dept. of Internal Medicine and Therapeutics, Fondazione IRCCS Policlinico San Matteo, Pavia, Italy), Elena Torri (Bresciamed, Bresciamed, Brescia, Italy), Federico Mento (Dept. of Information and Commun. Eng., Univ. of Trento, Trento, Italy), Domenico Milardi (Dept. of Medical and Surgical Sci., Fondazione Policlinico Universitario Agostino Gemelli IRCCS, Rome, Italy), Paola Del Giacomo (UOC Malattie Infettive, Dipartimento Scienze di Laboratorio e Infettivologiche, Fondazione Policlinico Universitario Agostino Gemelli IRCCS, Rome, Italy), Giuseppe De Matteis, Maria Livia Burzo (Dept. of Medical and Surgical Sci., Fondazione Policlinico Universitario Agostino Gemelli IRCCS, Rome, Italy), Anna Rita Larici (Dept. of Diagnostic Imaging, Oncological Radiotherapy and Hematology, Fondazione Policlinico Universitario Agostino Gemelli IRCCS, Rome, Italy), Maurizio Pompili (Dept. of Medical and Surgical Sci., Fondazione Policlinico Universitario Agostino Gemelli IRCCS, Rome, Italy), Libertario Demi (Information Eng. and Comput. Sci., Univ. of Trento, Trento, Italia, Italy), and Riccardo Inchingolo (Pulmonary Medicine Unit, Dept. of Medical and Surgical Sci., Fondazione Policlinico Universitario Agostino Gemelli IRCCS, Gemelli, Italy)

Lung ultrasound (LUS) has been reported as a useful tool to intercept lung peripheral changes (LPC) in COVID-19 pneumonia. Sixteen confirmed COVID-19 pneumonia patients underwent LUS using a standard sequence of scans in 14 landmarks. A score ranging from 0 to 3, according to Soldati's proposal, was reported for each landmark. High-resolution CT-scan of the chest (HRCT) was performed within 48 h prior to or after LUS. For each corresponding HRCT area, was reported a score (0 normal peripheral lung, 1 minimal LPC, 2 peripheral ground glass opacities (GGOs), 3 peripheral lung consolidations with or without GGOs) LUS showed sensitivity 92.1%, specificity 90%, PPV 96.8% to intercept LPC on HRCT (scores $\neq 0$). Higher LUS scores (2–3), corresponding to worst changes, showed sensitivity 70.1%, specificity 84%, PPV 78.1% to intercept higher HTCT scores (2–3). The overall score, for both LUS and HRCT, over 14 landmarks, showed no significant differences (paired t-test $p = 0.055$). An overall score $\geq 24$ was reported in five cases by LUS and 6 cases by HRCT. No significant differences also for patients either with more than three landmarks with score 3 or with 8 landmarks out of 14 with score 2–3 ($p = 0.16$). LUS showed good sensitivities and specificities compared to HRCT.

**12:00**

**4aBAb3. Is lung ultrasound a predictor of worsening in Covid-19 patients?** Umberto Sabatini (Internal Medicine, Policlinico San Matteo, Pavia, Piazzale Golgi 1, Pavia 27100, Italy, sabatini.u@gmail.com), Lucia Padovini, Gianluca Lettieri, Anita Fiengo, Giulia Gori, Federica Lepore, Matteo Garolfi, and Tiziano Perrone (Internal Medicine, Policlinico San Matteo, Italia, Italy)

SARS-CoV-2 infection can generate different responses in patients, ranging from asymptomatic virus shedding to severe pneumonia associated with high mortality. To evaluate the potential prognostic role of a recently introduced Lung Ultrasound (LUS) protocol in this context, a cohort of 52 consecutive laboratory-confirmed COVID-19 patients underwent LUS examination upon the admission and before the discharge in an Internal Medicine ward. LUS score was derived from 14 body-landmarks (2 anterior, 2 lateral and 3 posterior per hemithorax). Specific scores were assigned depending on the sole presence of horizontal artifacts (0), the presence of isolated vertical artifact only (1), the evidence of sub-pleural consolidations (2), the presence of confluent vertical artifact and/or confluent/large consolidations (3). We then investigated the association between the total LUS score severity and worsening, defined as a combination of high flow oxygen support, intensive care unit admission, or 30-day mortality as primary endpoint. Preliminary results of the study show that worsening outcome was reached by 20 (39%) patients during the observation period; average LUS score was 20.4 (SD 8.4) and 29.2 (SD 7.3) in patients without and with worsening, respectively. At univariable analysis, the total LUS score at admission was associated with higher odds of worsening.

**12:20**

**4aBAb4. The impact of B-lines' frequency characterization on lung ultrasound imaging, *in vitro*- and *in vivo* study.** Federico Mento (Dept. of Information and Commun. Eng., Univ. of Trento, Trento, Italy, federico.mento@unitn.it), Gino Soldati (Diagnostic and Interventional Ultrasound Unit, Valle del Serchio General Hospital, Lucca, Italy), Renato Prediletto (Inst. of Clinical Physiol., National Res. Council, Pisa, Italy), Marcello Demi (Fondazione Toscana Gabriele Monasterio, Pisa, Italy), and Libertario Demi (Dept. of Information and Commun. Eng., Univ. of Trento, Trento, Italia, Italy)

Lung ultrasound (LUS) is an established technique that allows clinicians to evaluate the state of the lung surface. However, LUS is based on the visual evaluation of imaging artifacts (e.g., A- and B-lines) and is hence affected by qualitative and subjective interpretations. Moreover, the impact of imaging settings on the visualization of artifacts is still greatly unknown. To shed light on these fundamental aspects of LUS, the results from two studies are presented. The first analyses through controlled *in vitro*- experiments the dependence of B-lines on three key imaging parameters, viz., center frequency, focal point, and active-aperture size. The second investigates thanks to a clinical study involving 26 patients the potentiality of B-line frequency characterization to differentiate pulmonary fibrosis from other diseases. A multi frequency imaging approach was used. Center frequencies ranging from 3 to 6 MHz and from 3 to 12 MHz were investigated for the *in vivo* and *in vitro*- study, respectively. Raw RF data were acquired with a ULA-OP platform and a LA533 linear array probe. Results show that the imaging frequency significantly influenced the intensity of B-lines and that their frequency characterization can be exploited to discriminate fibrotic patients with a sensitivity and specificity equal to 92%.

## Session 4aCAa

# Computational Acoustics: General Topics in Computational Acoustics I

Sheri Martinelli, Cochair
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Samuel F. Potter, Chair
*Computer Science, University of Maryland, 8125 Paint Branch Dr., College Park, MD 20742*

**Chair's Introduction—9:30**

## *Contributed Papers*

**9:35**

**4aCAa1. Wave propagation in a one-dimensional bar with rate-independent hysteresis.** Pravinkumar R. Ghodake (Mech. Eng., Indian Inst. of Technol., Bombay, B-423, Hostel 14, IIT Bombay, Mumbai, Maharashtra 400076, India, mech7pkumar@gmail.com) and Saurabh Biswas (Mech. Eng., Indian Inst. of Technol. Jammu, Jammu, Jammu & Kashmir, India)

A single frequency ultrasonic wave propagation in nonlinear materials generates higher harmonics. The generation of higher harmonics depends on the type of material nonlinearity considered, like odd and even harmonics in quadratically nonlinear materials, and only odd harmonics in cubically nonlinear material. The objective of this study is to present numerical study of one dimensional wave propagation in rate-independent hysteretic media by considering one dimensional chain of spring masses. Preisach-Mayergoyz (1994) and Hodgdon hysteretic models (1988) are commonly used in the theoretical and numerical study of nonlinear wave propagation studies. Here we implement the famous scalar Bouc-Wen model (1976) and a recently developed two-state hysteresis model by Biswas and Chatterjee (2015) to consider hysteresis in the large spring mass system. Number of spring masses are decided based on the spatial resolution needed for the nonlinear wave propagation to capture higher harmonics. The study presents interesting comparison between the wave propagation through an aluminum bar damaged under low cycle fatigue modeled as two different rate-independent hysteretic models. Both models show higher harmonics in responses. However, being a scalar model, the Bouc-Wen model cannot capture small minor loops due to partial reversals, whereas the two-state model captures the minor loops well.

**9:55**

**4aCAa2. Numerical geometric acoustics.** Samuel F. Potter (Comput. Sci., Univ. of Maryland, 8125 Paint Branch Dr., College Park, MD 20742, sfpotter@umd.edu), Maria Cameron (Mathematics, Univ. of Maryland, College Park, MD), and Ramani Duraiswami (Comput. Sci., Univ. of Maryland, College Park, MD)

We present a new approach to offline room acoustic modeling and simulation based on solving the eikonal equation, a nonlinear partial differential equation describing the arrival times of a high-frequency wavefront. We call the approach *numerical geometric acoustics*, since it combines some of the advantages of both numerical acoustics, where the acoustic wave equation or the Helmholtz equation are solved, and geometric acoustics, typically based on a variety of raytracing techniques. In our approach, reflection and diffraction effects are determined from boundary conditions connecting different eikonal problems generated by a recursion. This allows us to compute the *multipath eikonal*, a multivalued function parametrizing the arrival times of all acoustic rays propagating throughout a scene. Each branch of the multipath eikonal is computed by numerically solving the eikonal equation using a recently developed high-order semi-Lagrangian direct solver. This approach naturally encompasses spatially varying sound speeds. Our solver is compact and sufficiently high-order to allow us to transport the amplitude prefactor directly using paraxial raytracing. We discuss how to compute the acoustic parameter fields typical of precomputed room acoustic simulations used for virtual reality or games.

**10:15**

**4aCAa3. Using a linear microphone array to evaluate inhomogeneous sound-fields of recreated enclosures.** Evan Chertok (Architecture, Rensselaer Polytechnic Inst., 42 Forestdale Rd., Kinnelon, NJ 07405, echertok@optonline.net), Jonathan Mathews (Architectural Sci. - Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY), Samuel Chabot (Rensselaer Polytechnic Inst., Troy, NY), and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

Auditory Virtual Environments (AVE) typically recreate the sound at a specific listener position. Extended loudspeaker-based systems typically use the rich capabilities to enlarge the sweet spot, so a broader audience can experience the sound from the same virtual listener position. An alternative approach is taken with Rensselaer's Collaborative-Research Augmented Immersive Virtual Environment Laboratory (CRAIVE-Lab). Using 128 horizontally arranged loudspeakers, inhomogeneous sound fields are produced over a user area of $10 \times 12$ sqm, recreating the sound of acoustical enclosures using a ray-tracing method. A linear microphone array, consisting of 40 omnidirectional microphones, spaced 15 cm apart, is used to measure the fidelity of the reproduced inhomogeneous sound fields. This is accomplished by comparing the spatial room impulse responses measured in the CRAIVE-Lab to those generated synthetically (or computationally). The arrival times of direct sound and reflections are compared for both simulation methods. The analysis focuses on the spatial properties of the arriving wave fronts, including the direction of arrival and curvature, to determine the degree of accuracy to which the inhomogeneous properties of the sound field are reproduced. [Work supported by NSF #1909229.]

**10:35**

**4aCAa4. Simulations of room acoustics using fast multipole boundary element methods.** Nail Gumerov (Inst. for Adv. Comput. Studies, Univ. of Maryland at College Park, 4228 Iribe Ctr., College Park, MD 20742, ngumerov@umd.edu) and Ramani Duraiswami (Inst. for Adv. Comput. Studies, Univ. of Maryland at College Park, College Park, MD)

Room impulse responses can be simulated using various methods that differ in their computational complexity and accuracy. While boundary element methods (BEM) potentially can provide accurate simulations in

complex geometries including effects such as diffraction and absorption, their use is limited due to relatively large values of parameter kD (k is the wavenumber and D is the room diameter). The size of the mesh also grows as the square of this parameter. We show that the fast multipole (FMM) accelerated BEM realized on multicore workstations can handle problems for rooms of $D \sim 10$ m for the frequency range up to ~5 kHz ($kD \sim 1000$). For such problems, needing meshes with several million elements, stable and efficient methods for high frequency FMM are needed. We discuss these issues and demonstrate results of such computations both in frequency and time domains. In simple cases numerical solutions are compared with analytical solutions obtained by the method of images. For more complex cases including rooms of different shapes with baffles and openings two methods, the direct and indirect BEM, are studied and their performance discussed. [This work is supported by VisiSonics Corporation.]

THURSDAY MORNING, 10 DECEMBER 2020 — 11:15 A.M. TO 12:45 P.M.

## Session 4aCAb

## Computational Acoustics: General Topics in Computational Acoustics II

Amanda Hanford, Cochair
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Song Wang, Cochair
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**Chair's Introduction—11:15**

### *Contributed Papers*

**11:20**

**4aCAb1. Effect of flow on the acoustic length correction factor of a Helmholtz resonator neck at high Strouhal number: A symmetric three-dimensional numerical parametric study.** Diego M. Tuozzo (Acoust. and Vib. Lab., Federal Univ. of Santa Catarina, Campus Universitario Trindade, Centro Tecnologico, Florianopolis, Santa Catarina 88040900, Brazil, dmtuozzo@mopt.com.br), Arcanjo Lenzi (Acoust. and Vib. Lab., Federal Univ. of Santa Catarina, Florianópolis, Santa Catarina, Brazil), and Olavo M. Silva (Multidisciplinary Optimization Group, Federal Univ. of Santa Catarina, Florianópolis, Brazil)

The effect of flow on the acoustic length correction factor (ALCF) of a Helmholtz resonator (HR) neck is investigated numerically in order to achieve an expression to calculate the respective flow-acoustic length correction factor (FALCF) as a function dependent on the ratio between the radii of the neck and cavity (up to 0.4) and on the Mach (Ma) number (up to 0.1). The ALCF is of great interest in one-dimensional acoustic applications for achieving better prediction of local effects. In this work, the effects of turbulent flow and radial and axial neck-cavity wave motions are added to improve the one-dimensional HR's resonant frequency prediction model. A symmetric three-dimensional HR model is parameterized and adopted to solve a set of CFD problems (RANS equation and turbulent SST model), with different geometry parameters. The acoustic fluid is air at 20 °C and is considered incompressible. The predictions of the numerical model are validated with experimental studies available in the literature. Different formulations employed to predict the resonance frequency of an HR for the $Ma = 0$ case are investigated and compared to CFD results as a way to verify its prediction capability related with the HR's geometry. Also, the possibility to obtain the FALCF factor from the neck's acoustic impedance is investigated and compared with the expression derived from the HR's resonance frequency.

**11:40**

**4aCAb2. Computational aeroacoustics for low Mach number flow using the lattice Boltzmann method.** Song Wang (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol. (CIRMMT), Schulich School of Music, McGill Univ., 555 Rue Sherbrooke Ouest, Montreal, QC H3A 1E3, Canada, song.wang5@mail.mcgill.ca) and Gary Scavone (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol. (CIRMMT), Schulich School of Music, McGill Univ., Montreal, QC, Canada)

Computational aeroacoustics at low Mach number is challenging because of the comparative order of magnitudes between the acoustic perturbations and the numerical errors. The lattice Boltzmann method has been proven to be advantageous for aeroacoustics with a lower dissipation. The recursive and regularized LBM scheme (LBM-rrBGK), which has only been developed recently and tested in aeroacoustic problems, shows advantages over the traditional LBM-BGK. In this paper, results of computational aeroacoustics for low Mach number flow using rrBGK are presented. Two benchmark problems are tested, including the flow passing around a circular cylinder and the sound radiation of a cylindrical duct with uniform flow. The results are compared with either simulation, experiment or analytical results in the literature, and show a good agreement. Finally, the sound radiation directivity of a horn in the presence of mean flow is studied.

12:00

**4aCAb3. Direct numerical simulation and parametric study of the noise generated from particle dispersion in decaying homogeneous isotropic turbulent flow.** Wei Wang (Mech. and Aerosp. Eng., Univ. of Florida, Gainesville, FL 32611, wei.wang@ufl.edu) and Steven A. Miller (Mech. and Aerosp. Eng., Univ. of Florida, Gainesville, FL)

The main objective of this study is to understand the noise produced by the dispersion of particles in a simple turbulent flow and to classify the statistics of the particles that influence the noise. The three-dimensional, time-dependent, non-stationary flow-fields of the homogenous isotropic turbulence with microscale Reynold numbers varying from 15 to 50 are computed using direct numerical simulation. 1, 5, 10, 50, and 100 thousand solid particles of varying diameters and densities are randomly injected into the computational domain. The particles' positions, velocities, and temperatures are assessed by integrating the equations of particle motion along each trajectory. The acoustic pressure time history at different far-field observer locations are predicted using the Crighton and Ffowcs Williams (CFW) acoustic analogy for two-phase turbulence. The root mean square acoustic pressures at different observer locations are ensemble-averaged and compared. Three distinct sources of noise from the CFW acoustic analogy are computed, and the effects of particle dispersion on noise sources are presented. The root mean square acoustic pressures increase drastically with an increasing number of particles in the cases of large particles, while the increases in the cases of small particles are not significant comparing to the single-phase flow.

12:20

**4aCAb4. An arbitrary-order immersed finite-difference method for wave propagation through fluid/solid interfaces.** Roberto Sabatini (Ctr. for Maritime Res. and Experimentation, Viale S. Bartolomeo, La Spezia 19126, Italy, roberto87sabatini@gmail.com), Yan Pailhas (Ctr. for Maritime Res. and Experimentation, La Spezia, Italy), Paul Cristini (Laboratoire de Mecanique et d'Acoustique, Marseille, France), and Angeliki Xenaki (Ctr. for Maritime Res. and Experimentation, La Spezia, Italy)

The immersed interface method (IIM) has recently gained popularity for the resolution of the linearized equations of continuum mechanics (LECM) that model the propagation of acoustic and elastic waves in complex ocean environments. In the IIM, the boundaries between media, e.g., the interface between the ocean and the seabed, are considered immersed in a regular Cartesian grid. Far from the interfaces, classical finite differences are employed. Conversely, at points near the boundaries, the standard schemes are modified for the solution to satisfy both the LECM and the required interface conditions. Current implementations of the IIM are generally second-order accurate. However, high-quality simulations require a better resolution in terms of points-per-wavelength, especially when the number of grid points in the computational domain is limited (e.g., by the GPU memory). In this presentation, an arbitrary-order immersed finite-difference method is proposed. The improvement of the solution accuracy when increasing the order of the numerical scheme is more particularly demonstrated on scattering problems relevant to the mine-countermeasures community. Finally, the proposed IIM is compared with the widely employed spectral-element method.

THURSDAY MORNING, 10 DECEMBER 2020 9:30 A.M. TO 11:00 A.M.

## Session 4aEA

## Engineering Acoustics, Physical Acoustics, Structural Acoustics and Vibration, and Underwater Acoustics: Advanced Materials for Acoustic Transducers

Thomas E. Blanford, Cochair
*The Pennsylvania State University, State College, PA 16804*

Michael R. Haberman, Cochair
*Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758*

4a THU. AM

**Chair's Introduction—9:30**

***Contributed Papers***

9:35

**4aEA1. Subjective evaluation of carbon nanotube thermophones using spoken text.** Troy Bouman (Mech. Eng. - Eng. Mech., Michigan Technol. Univ., 815 R.L. Smith MEEM Bldg., 1400 Townsend Dr., Houghton, MI 49931, tmbouman@mtu.edu) and Andrew R. Barnard (Mech. Eng. - Eng. Mech., Michigan Technol. Univ., Houghton, MI)

Carbon nanotube (CNT) loudspeakers are modern thermoacoustic speakers. Their extreme light weight, lack of moving parts, and geometric flexibility provide the engineering community an exciting new tool in their sound generation toolbox. Prior to this study, no effort has been made to understand their subjective performance. All data in literature are objective metrics, e.g., sound pressure, sound power, total harmonic distortion, etc. This effort uses a hybrid paired comparison and modified rhyme test jury study to compare multiple CNT loudspeaker drive signal processing methods to traditional moving coil loudspeakers. The study scope was limited to single word spoke text.

**9:55**

**4aEA2. Performance analysis of an enclosed, coaxial carbon nanotube (CNT) speaker in presence of flow using COMSOL Multiphysics.** Suraj Prabhu (Mech. Eng., Michigan Technol. Univ., 815 R.L. Smith MEEM Bldg., 1400 Townsend Dr., Houghton, MI 49931, smprabhu@mtu.edu) and Andrew R. Barnard (Mech. Eng. - Eng. Mech., Michigan Technolog. Univ., Houghton, MI)

Carbon Nanotube (CNT) speakers are solid state speakers that operate on the thermovicsous acoustic principle. These speakers generate sound proportional to the oscillations in the surface temperature of the CNT film on application of an alternating current. CNT films are flexible, stretchable, transparent and extremely lightweight. These properties make it possible to manufacture the CNT speakers in a variety of shapes such as cylindrical, spherical, etc. An enclosed, coaxial CNT speaker was designed for noise cancellation in exhausts, ducts, etc. The enclosed design was simulated using COMSOL Multiphysics without the presence of flow in the medium of operation. In this study, simulations are carried out by introducing flow in the medium, particularly with different Mach numbers. The turbulent flow model results of a simple pipe are compared with the results obtained from Fluent to verify the COMSOL Multiphysics model. Then an enclosed, coaxial CNT speaker model is simulated in COMSOL Multiphysics by mapping the CFD mesh parameters on an acoustic mesh. The results obtained are compared with the simulation results from no-flow condition. This study helped to understand the effect of different Mach number turbulent flows on the sound pressure level generated by the enclosed, coaxial CNT speaker.

**10:15**

**4aEA3. Flexible electrostatic transducer with tuned acoustic impedance for improved sensing of body-and water-borne sounds.** Ian M. McLane (Elec. and Comput. Eng., Johns Hopkins Univ., 3400 North Charles, Barton Hall 105, Baltimore, CA 21218, imclane1@jhu.edu), Valerie Rennoll, Adebayo Eisape, Mounya Elhilali, and James West (Elec. and Comput. Eng., Johns Hopkins Univ., Baltimore, MD)

Acoustic transducers that are designed for airborne sound suffer from impedance mismatches when used in different media. These mismatches result in lowered energy density at the transducer and increased corruption by external noises. Coupling layers can mitigate mismatches, but restrict real-world applicability through limited bandwidth, thickness- dependent sensitivity, and bonding issues. We introduce a self-powered, flexible, electrostatic transducer with a highly tunable acoustic impedance and wide frequency bandwidth, without matching layers. The transducer can be precisely tuned to impedance values in the range of 1 to 2.5 MRayls, which includes skin, fresh and saltwater, and most plastics. Metal-coated microstructures on the surface of an elastomer create small gaps with the charged polymer electret film that are compressed by mechanical vibrations in the elastomer and induce a change in the voltage output. The transducer exhibits a sensitivity greater than 2V/N and an SNR improvement of 35 dB in highly noisy environments. Major factors that affect the output of the transducer are analyzed, including the dimensions of the microstructures, elastomer thickness, and properties of the electret polymer. The acoustic properties, such as sensitivity and improved rejection of environmental noise, are experimentally evaluated through a body phantom in simulated noise environments in a semi-anechoic chamber.

**10:35**

**4aEA4. A heat balance model to explain thresholds for thermoacoustic sound production in seawater using metal electrodes.** Michael McBeth (Naval Information Warfare Ctr. Atlantic, NASA Langley Res. Ctr., 8 North Dryden St., M.S. 473, Hampton, VA 23681, michael.s.mcbeth@navy.mil)

Experiments conducted with solid metal wire electrodes in seawater generated second harmonic sound waves thought to be thermoacoustic in origin. Using sine wave bursts of 3 to 25 cycles of 10 kHz voltage across the electrodes, the applied voltage threshold for thermoacoustic sound production was observed to decrease almost linearly for bursts up to ten cycles. At the time of these experiments, around 2014, we were unable to explain these sound production threshold values. A mass density continuity equation with a negative source term is used to explore the observed threshold values. The negative source term represents the decrease in fluid density with each half cycle of applied voltage due to the ohmic heating. This approach builds on an earlier heat balance model that successfully explains an observed delay in thermoacoustic sound production. Model results are compared with the experimental observations.

## Session 4aEDa

# Education in Acoustics: General Topics in Acoustics Education

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

**Chair's Introduction—9:30**

## *Contributed Papers*

**9:35**

**4aEDa1. Last call for the Wiki 4 Year of Sound 2020 campaign and the contributions from students.** Thais C. Morata (Div. of Field Studies and Eng., NIOSH, 1090 Columbia Tusculum Ave., Cincinnati, OH 45229, tmorata@cdc.gov)

The International Year of Sound (IYS 2020) is a global initiative to highlight the importance of sound in all aspects of life. Wikipedia is an important channel to share science information with the public. The National Institute for Occupational Safety and Health partnered with several acoustical societies in organizing an online event to facilitate the improvement of Wikipedia content related to sound. The Wiki4YearOfSound2020 platform provided guidance on how to identify topics, improve, or translate existing articles. Throughout 2020, the Wikimedia outreach dashboard has been tracking progress, individual contributions, and their results, which reached tens of millions of views. A few educational programs tasked students to contribute high-quality content to Wikipedia as part of their class assignments. They used different dashboards to track progress. Students learned about the process and the value of translating science to the public. The goal of learning a topic well enough to be able to write in a style accessible to lay readers is challenging and requires more research than a traditional assignment. Finally, metrics made available by Wikipedia platforms were considered motivating and rewarding by instructors and students.

**9:55**

**4aEDa2. SpeakEasy: Browser-based automated multi-language intonation training.** Jeramey Tyler (Cognit. Sci., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, tylerj2@rpi.edu), Mei Si (Cognit. Sci., Rensselaer Polytechnic Inst., Troy, NY), Helen Zhou (Communications and Media, Rensselaer Polytechnic Inst., Troy, NY), hui su (CISL, Rensselaer Polytechnic Inst., Troy, NY), and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

Learning to speak a second language involves a cycle of observation, mimicry, and feedback. A student observes a teacher, attempts to copy the teacher's performance, and then the teacher provides feedback on how the student performed. When the students' access to feedback is limited, so is their ability reinforce their learning. In this work a web based application, Speakeasy tools, in introduced to provide remote students with automated visual intonation feedback for multiple languages. For this study, participants are selected from the pool of Speakeasy users and their interactions with the application are observed over a set period of time. The application presents participants with native speaker examples generated via text-to-speech in the form of audio samples, fundamental frequency visualizations, and grapheme and phoneme level timelines. Participants are able to record and review an unlimited number of practice attempts which are processed using the same pipeline used for the native speaker examples. Using the Root-Mean-Square Error (RMSE) between native and participant fundamental frequencies over time, the practice attempt is assigned a score. Practice-attempt scores with respect to time spent using the application provide a metric for measuring a participant's progress. [Work supported by RPI Seed Grant and CISL.]

**10:15**

**4aEDa3. Ogene Bunch music analyzed through the visualization and sonification of beat-class theory with ski-hill and cyclic graphs.** Andrea Calilhanna (MARCS Inst. for Brain, Behaviour and Development, Bullecourt Ave. Milperra, Sydney, New South Wales 2214, Australia, A.Calilhanna@westernsydney.edu.au)

Traditional Ogene music (metal gong) is a music genre originating in the eastern communities of Nigeria which retains an intrinsic communicative function and music style of Igboland characterized by call and response vocals, polyrhythms and improvisation. Through applying a psychoacoustic approach using visualizations and sonifications of beat-class theory with ski-hill and cyclic graphs, a listener is enabled to represent their experience of both music and mathematics embodied acoustics. This paper demonstrates the evolution of Western music theory to have the capacity to represent rhythm, meter, and pitch by visualizing beat-class theory through ski-hill and cyclic graphs (mathematical music theory). This new approach is suitable for understanding Igbo music which, like the majority of the world's music, is transmitted aurally and not notated. Applying a psychoacoustic approach, this paper illustrates how Igbo music can be accurately preserved for future generations and included in the scholarly discussion. The paper also reveals structural details of the music which may not be accounted for previously through the use of traditional Western music theory.

**10:35**

**4aEDa4. Interdisplinary Physical Music: A Blind Spot in Education on Acoustics.** Stephen G. Onwubiko (Music, Univ. of Nigeria, Nsukka, Enugu 234042, Nigeria, stephen.onwubiko@gmail.com) and Andrea Calilhanna (MARCS Inst. for Brain, Behaviour and Development, Sydney, New South Wales, Australia)

In attaining a degree in music, education on musical acoustics is multidisciplinary. Musical acoustics contains specialized knowledge, history, methodology, and practices, yet music itself is interdisciplinary. Ideally, acousticians recognize and validate that music is scientific and so, turn a blind eye or pay little attention to interdisciplinary physical music (i.e., migrating from music to musical acoustics). This paper examines the critical fault line erupting beneath the structural foundations of music through musical acoustics to navigate questions seldomly asked. These questions include–who should teach interdisciplinary physical music, e.g., music and mathematics, musical acoustics, psychoacoustics, what are the sets of curriculum standards and evaluation procedures suitable for the inter-multidisciplinary physical music. Finally, the paper proposes the reason why these paths are seldom crossed and provides a solution: a psychoacoustic approach to mathematical music theory as conduit to the intersection of fields with concepts traversing many disciplines.

## Session 4aEDb

# Education in Acoustics: Undergraduate Research Symposium Poster Session

Authors will be at their posters from 11:15 a.m. to 12:00 noon.

### *Contributed Papers*

**4aEDb1. Acoustic conditions in occupied K–12 classrooms as measured over four continuous weekdays.** Drake A. Hintz (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 9431 U Plaza #8, Omaha, NE 68127, Drake.a.hintz@gmail.com) and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE)

A recent study at the University of Nebraska–Lincoln involved capturing sound level data over four continuous weekdays in 55 K-12 classrooms during the 2017–2018 academic year. Equivalent sound levels in each classroom were logged every 10 seconds with an integration time of 10 s over two different weeks: once during heating season and once during cooling season. Among the results presented are the measured classroom reverberation times, average sound levels in each classroom when speech was occurring and when speech was not occurring, the percent of time that levels in each classroom exceeded certain values, and the daily variation of these metrics. The data gathered from this study are compared to data from 220 K-12 classrooms gathered in the 2015–2016 and 2016–2017 academic years. [Work supported by the United States Environmental Protection Agency Grant No. R835633.]

**4aEDb2. Acoustical demonstration of slurring characteristics using a French horn.** Tiffany A. Bixler (Phys. Dept., U.S. Naval Acad., Annapolis, MD 21402, tiffanybixler7@gmail.com) and Murray S. Korman (Phys., U.S. Naval Acad., Annapolis, MD)

In this experiment, we will be attempting to re-create Shona Marie Logie's slurring experiments where a Conn 8D (modern wide bore horn), a Boosey and Hawkes horn, and a trumpet were the selected brass instruments. [S. M. Logie, "An acoustical study of the playing characteristics of brass wind instruments," Ph.D. thesis (University of Edinburgh, 2012)]. However, instead of the instruments mentioned above, experiments will be performed using an "F" horn. Our goal is to inform students about musical acoustics and slurring by doing some fundamental experiments. When conducting the experiments (in two parts), we perform (1) forced oscillation slurring experiments where the mouthpiece is driven by a loudspeaker and utilize two different microphones (located near the mouthpiece and near the bell) and (2) conduct experiments where the performer is slurring using an excerpt of a musical arrangement. Experiments will be conducted in a soundproof room and an anechoic chamber. We will be using an excerpt from Turandot, by Giacomo Puccini and arranged (for French horn) by Robert van Beringen, called Nessun Dorma (meaning "Let no one sleep").

**4aEDb3. Circular synthetic aperture ultrasonic acoustic imaging using a 2.25 MHz underwater sonar.** Ava B. Twitty (Phys. Dept. U.S. Naval Acad., Annapolis, MD 21402, avat1226@gmail.com) and Murray S. Korman (Phys., U.S. Naval Acad., Annapolis, MD)

Experiments were performed in the USNA physical acoustics lab to demonstrate underwater sound imaging using circular synthetic aperture acoustic C-SAA imaging in two dimensions. The apparatus includes a single 2.25 MHz submersion transducer acting as both transmitter and receiver (2.54 cm diam with a manufacture's specified 4 cycle pulse shape) located 15 cm from the target's center. A structure of thin perforated disks (7.6 cm diameter) comprise a step-motor controlled rotational platform that supports several slender stainless steel cylindrical rod vertical targets (2 mm diameter, 10 cm long). The apparatus and supporting electronics serves in part as a teaching tool for imaging targets in a laboratory environment. This setup might be called inverse C-SAA considering a medical imaging tomography arrangement where the transmitter-receiver is fixed while the targets rotate at small angular intervals. Here, the platform "stops" for an echo versus time measurement and then rotates (or "hops") to the next location for the following measurement. The transmitted pulse is generated by a rectangular pulse (duration $T = 0.5/2.25$ MHz) or utilizing a short tone burst. Correlation pulse compression can be used if a chirp is transmitted. A back-projection algorithm performed on Mathematica® was used to generate a 2-D reflectance image—for several targets.

**4aEDb4. Spectral analysis of turbulent boundary layer pressure fluctuations collected by a MEMS microphone array.** Ethan Saff (Lexington Christian Acad., Lexington, MA) and Robert D. White (Mech. Eng., Tufts Univ., 200 College Ave., Medford, MA 02155, r.white@tufts.edu)

A MEMS-based microphone array was previously constructed to collect and measure pressure fluctuations due to turbulence under flat plate turbulent boundary layers with high spatial resolution (≈1.3 mm). Data were collected from this array at Mach numbers up to 0.6 and Reynold's numbers up to $10^7$ per meter in three flow facilities, at NASA Ames, the Univ. of Toronto, and Spirit Aerosystems. The measured pressure fluctuations are being processed to identify the turbulence characteristics of the data. Comparisons of the single point power spectral density to previously existing models, such as those of Chase and Goody, as well as wave speed estimates derived from the phase slope, suggest that turbulent characteristics are significantly present in the measured data. In a variety of different cases, the data will be compared to the Corcos model for cross spectral density in streamwise, spanwise, and intermediate directions to further confirm the data's validity and to quantitatively investigate the validity of the separable model used in Corcos's expression.

**4aEDb5. Characterization of sand moisture profiles for improved atmospheric acoustic modeling.** Faith A. Cobb (Eng., East Carolina Univ., Slay Hall 248, Greenville, NC 27858, cobbf18@students.ecu.edu), Nia Wilson (Eng., East Carolina Univ., Greenville, NC), Andrea Vecchiotti, Diego Turo, Joseph Vignola (Mech. Eng., The Catholic Univ. of America, Washington, DC), and Teresa J. Ryan (Eng., East Carolina Univ., Greenville, NC)

This work presents an effort to understand and characterize the role moisture content plays in how sound interacts with sand on the beach. This work is part of a larger project developing a numerical model of long-range (~3 km) atmospheric acoustic propagation in littoral or riverine environments with a near-shore acoustic source and on-shore receivers. Previous experiments have studied the effects of moisture content of sand samples taken from various distances from the water's edge of a man-made lake with an artificial beach. This works seeks to perform the same studies in natural coastal areas by recording moisture content on site via a moisture probe at several depths, collecting samples to determine gravimetric water content of a bulk sample, and conducting testing with an impedance tube. Samples will be collected at various distances from the shoreline in order to inform a related effort that seeks to capture the acoustic characteristics of the sandy beach for use numerical modeling.

**4aEDb6. Comparison of existing and experimental weather models of the near surface atmospheric boundary layer.** Nia Wilson (East Carolina Univ., Greenville, NC, wilsonni18@students.ecu.edu), Faith A. Cobb (East Carolina Univ., Greenville, NC), Diego Turo, Joseph Vignola (Mech. Eng., The Catholic Univ. of America, Washington, DC), and Teresa J. Ryan (Eng., East Carolina Univ., Greenville, NC)

This work presents a comparison between existing weather models and experimental measurements of the lower atmospheric boundary layer. This work is part of a larger project developing a numerical model of long-range (~3 km) atmospheric acoustic propagation in littoral or riverine environments with a near-shore acoustic source and on-shore receivers. Temperature, humidity, and pressure data are collected with the use of iMet-XQ2 sensors mounted on an unmanned aerial vehicle. Vertical measured profiles that will then be compared to existing common models and assumptions (such as Monin-Obukhov similarity theory) to gain a better understanding of these mathematical models and how sufficient these models are at representing the meteorological parameters that will affect acoustic transmission.

THURSDAY MORNING, 10 DECEMBER 2020 — 9:30 A.M. TO 11:00 A.M.

## Session 4aMUa

## Musical Acoustics: General Topics in Musical Acoustics IV

Kurt R. Hoffman, Cochair
*Physics, Whitman College, 345 Boyer Ave., Hall of Science, Walla Walla, WA 99362*

Mark Rau, Cochair
*Music, Stanford University, 660 Lomita Court, Stanford, CA 94305*

**Chair's Introduction—9:30**

### *Contributed Papers*

**9:35**

**4aMUa1. Lyrics on the melody or melody of the lyrics?** Archi Banerjee (IIT Kharagpur, Kharagpur 721302, India, archibanerjee7@gmail.com), Souparno Roy (Jadavpur Univ., Kolkata, India), and Priyadarshi Patnaik (IIT Kharagpur, Kharagpur, India)

The musicians say "When a marriage happens between a lyric and a melody, only then a true song is born"! A proper song should express the true meaning of its lyrics through the movements of its melodic structure. "Which impacts more: Melody or lyrics?"—The answer is still unknown. The present work explores the acoustical contribution as well as the neuro-cognitive impact of lyrics in different genres of songs evoking contrast emotions. Four songs of two different genres (i.e., two songs from each genre depicting opposite/contrast emotions: joy-sorrow) were recorded from a trained musician who was asked to consecutively sing (with proper meaningful lyrics) and hum (without using any lyric or meaningful words) the songs, the obtained audio signals were later put to acoustical analysis. Later EEG signals were recorded for five naûve participants with different sets of song-humming (i.e., with lyric-without lyric) versions of the same melodic content as stimuli, to study their corresponding bio-signal (EEG) attributes. DFA technique was employed to calculate and compare the Hurst Exponent values for all of the recorded auditory signals as well as the EEG responses. This is a pilot study in the context of Indian music which endeavours to analyze the contribution of lyrics in songs of different genre as well as different emotional content in both acoustic and neuro-cognitive domain.

**9:55**

**4aMUa2. Where does the beat fall? Speech-beat alignment in Mandarin and English singing.** Cong Zhang (Univ. of Kent, Cornwallis North West, Canterbury, Canterbury CT2 7NF, United Kingdom, congzhang.oxford@gmail.com) and Charlotte A. Slocombe (Univ. of Kent, Canterbury, United Kingdom)

Text-to-sing generates singing from text input (i.e., music score with lyrics), from which only syllable-level speech-music alignment can be acquired. To enhance the text-to-sing models, more fine-grained phoneme-level information is needed. We therefore investigate the acoustic measurements of segments and their temporal relationship with music beats as an answer from a linguistics perspective. Two research questions are addressed: (1) Do beats align with syllable onsets or nuclear vowel onsets? (2) Do different types of consonants present different speech-beat alignment results? Unaccompanied singing by professional singers in two rhythmically dissimilar languages, English (15 songs) and Mandarin (25 songs), were analysed. Data were segmented manually into phonemes by a trained annotator; a music scholar independently labelled the beats. Preliminary results suggest that Mandarin songs strongly favour vowels as anchors for beats (66.7%) while only 52.9% of beats fall on vowels in English. Both languages show that the beats have a strong preference for the end of consonants and the beginning of vowels. Phoneme types also play a significant role in the speech-beat alignment distribution. Future modelling of the speech-beat alignment in singing and comparison with speech rhythm data will also contribute to linguistic rhythm theories.

10:15

**4aMUa3. Emotions from musical notes? A psycho-acoustic exploration with Indian classical music.** Shankha Sanyal (Lang. and Linguist, Jadavpur Univ., Kolkata, West Bengal 700032, India, ssanyal.sanyal2@gmail.com), Archi Banerjee (IIT Kharagpur, Kharagpur, India), Dipak Ghosh (Jadavpur Univ., Kolkata, India), and Samir Karmakar (Lang. and Linguist, Jadavpur Univ., Kolkata, India)

The most interesting feature of Indian Classical Music is the existence of Raagas. Each Raaga has its own peculiar ascending and descending movement called the Arohana and Avarohana. Even if two (or more) Raagas are made up of the same notes, the combinational varieties of notes evoke different emotions. In this work, we envisage to study how emotion perception in listeners' changes when there is an alteration of merely a single note in a pentatonic Raaga and also when a particular note(s) is replaced by its flat/sharp counterpart. 30-s recordings were done for two pair of Raagas which were chosen in a manner such that they are having difference in only one note keeping all others same. The fractal dimension of the auditory waveform provides a robust nonlinear quantitative parameter with which the two pair of audio clips can be compared. Also, the emotional appraisal from these two pairs were assessed on the basis of psychological listening tests as well as from cognitive response in the form of EEG experiments done on 10 participants. Interesting new results are obtained on how a trivial change in the note structure of a particular Raaga influences human emotion to a large extent.

10:35

**4aMUa4. Articulatory activity of the tongue, jaw, and lips during the second passaggio acoustic transition of female singers.** Richard C. Lissemore (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), Kevin D. Roon, and D. H. Whalen (Speech-Language-Hearing Sci., Graduate Center/The City Univ. of New York, New York, NY)

Sopranos typically exhibit an acoustic modification between 600 and 700 Hz (on /a/) during which second resonance ($R_2$) tracking of the second harmonic ($2f_o$) changes to first resonance ($R_1$) tracking of the fundamental ($f_o$). To quantify the alteration, the sound pressure level difference between the first two harmonics ($L_1$-$L_2$) was measured for chromatic scales sung between C5□ and G5□ (523 to 784 Hz) by 17 sopranos (9 judged by the first author as "techniqued" and 8 as "un-techniqued" and confirmed by perceptual experiment). Techniqued sopranos shifted from negative to positive values of $L_1$-$L_2$ as early as D5□ (587 Hz), while the least techniqued singers did not make the change at all. Articulatory correlates were measured using ultrasound of the tongue and optical tracking. Head-corrected tongue contours showed the most critical articulatory factor to be the size of a triangular area between two points on the hard palate and the most anterior tongue point. Sopranos who made the change earliest exhibited the largest anterior oral cavities whereas sopranos who never made the acoustic change had significantly smaller anterior cavities. Larger anterior oral cavities appear to accommodate lower frequencies of $R_2$, presumably lowering $L_2$ and increasing $L_1$-$L_2$. [Work supported by NIH grant DC-002717.]

THURSDAY MORNING, 10 DECEMBER 2020 — 11:15 A.M. TO 12:25 P.M.

## Session 4aMUb

## Musical Acoustics: General Topics in Musical Acoustics V

Kurt R. Hoffman, Cochair
*Physics, Whitman College, 345 Boyer Ave., Hall of Science, Walla Walla, WA 99362*

Mark Rau, Cochair
*Music, Stanford University, 660 Lomita Court, Stanford, CA 94305*

**Chair's Introduction—11:15**

### *Contributed Papers*

11:20

**4aMUb1. How much delay can you hear as a function of frequency? Towards the improved design of decorrelation algorithms.** Elliot K. Canfield-Dafilou (Ctr. for Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., 660 Lomita Ct, Stanford, CA 94305, kermit@ccrma.stanford.edu) and Takako Fujioka (Ctr. for Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., Stanford, CA)

The extent to which the human auditory system can perceive phase information from incoming sounds has been much debated. While some research has suggested that humans are extremely sensitive to phase distortion, some believe that small amounts of phase distortion—up to about 20 ms—are almost inaudible. The latter assumption, in particular, is related to signal processing techniques such as decorrelation algorithms, which are widely used for artificially controlling the perceived diffuseness of the source information. The present study systematically examined the audibility threshold of the delay confined in a band of frequencies using an adaptive (2-down, 1-up) 2AFC procedure. We tested two types of audio sample: foreign-speech and instrumental music. For each signal, we introduced delays in one of five frequency regions (center frequencies: 250, 570, 1000, 2160, and 4050 Hz; critical bandwidths). Listeners could hear delays on the order of 5–10 ms in frequency bands lower than 1000 Hz, while in high-frequency bands, the delay threshold was much lower, around 1.5–5 ms. Moreover, thresholds were general lower for speech than music. While most listeners reported that the distortions were subtle, the results highlight the importance of frequency-dependent, content-specific design of the temporal limits in decorrelation algorithms.

**11:40**

**4aMUb2. Quality of experience in musical acoustics.** Bozena Kostek (Audio Acoust. Lab., Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, bokostek@audioakustyka.org) and Magda Piotrowska (Hear Candy Mastering, Kraków, Poland)

Quality of sound is the most crucial issue in musical acoustics. This may be researched in the context of perceived experiences, the so-called *Quality-of-Experience* domain. For that purpose, subjective tests are carried out, the results of which, based on appropriate statistical analysis, serve as an indication of quality of sound or music. It should be pointed out that experience is not only subjective but also context-dependent. In addition to the external factors related to the test listening conditions, test method, reproduction devices, etc., a person's perception, experience, and cognitive state clearly play a critical role in the context. Despite significant advances in objective measuring methods, the only way to evaluate the subjective quality of musical sound or music is to prepare a set of tests in which a tester's opinion is acquired and then to quantify resulted preferences and choices. The problem that remains still open is mapping quantitative measurement results to the human's evaluation. The paper presents a method of mapping several subjective notions to objectively derived sound features. This is shown in the context of the subjective assessment of the mood of music. Correlation between vocabulary describing music mood and corresponding sound attributes is shown based on the listening tests performed.

**12:00**

**4aMUb3. Meta-learning for acoustic and music composing.** Robert B. Lisek (Ctr. of Adv. Studies, CEU, Bulwar Ikara 18/5, Wroclaw 54130, Poland, robertblisek@gmail.com)

We observe the success of artificial neural networks in simulating human performance on a number of tasks: such as image or sound recognition, natural language processing, etc. However, there are limits to state-of-the- art AI that separate it from human-like intelligence. Humans can learn a new skill without forgetting what they have already learned and they can improve their activity and gradually become better learners. Today's AI algorithms are limited in how much previous knowledge they are able to keep through each new training phase and how much they can reuse. In practice this means that it is necessary to build and adjust new algorithms to every new particular task. This is closer to a sophisticated data processing than to real intelligence. This is why research concerning generalisation are becoming increasingly important. A generalization in AI means that system can generate new compositions or find solutions for new tasks that are not present in the training corpus. Intelligent agent should have meta learning capabilities, should not just be able to memorize the solution to a fixed set of tasks during creating of stories, but learn how to generalize to new problems it encounters. The project proposes a solution for problems linked to organizing acoustic material and creation of new compositions based on meta-learning. Meta-Composer is a neural network equipped with the ability to combine acoustic materials and partial compositions in a flexible and combinatorial way to create a new consistent general composition.

**4aMUb4. Music recommendation based on acoustic features from the recording studio.** Tim Ziemer (Bremen Spatial Cognition Ctr., Univ. of Bremen, Enrique-Schmidt-Str. 5, Bremen 28359, Germany, ziemer@uni-bremen.de), Pattararat Kiattipadungkul (Faculty of Information and Commun. Technol., Mahidol Univ., Salaya, Thailand), and Tanyarin Karuchit (Faculty of Information and Commun. Technol., Mahidol Univ., Salaya, Thailand)

Producers of Electronic Dance Music (EDM) typically spend more time creating, shaping, mixing and mastering sounds, than with aspects of composition and arrangement. They analyze the sound by close listening and by leveraging audio monitoring tools, until they successfully created the desired sound character. DJs of EDM tend to play sets of songs that meet their sound ideal. We use audio monitoring tools from the recording studio to retrieve the sound ideal of the most popular DJs and perform a DJ classification, e.g., to predict "what your favorite DJ would play." The features include third-octave band VU, RMS and crest factor meters, phase scope, and the channel correlation coefficient. This new set of features and the focus on DJ sets is targeted at EDM as it takes the producer and DJ culture into account. With simple dimensionality reduction and machine learning these recording studio features enable us to attribute a song to a DJ with an accuracy of 63%. The features from the monitoring tools in the recording studio could serve for many applications in music information retrieval, such as genre, style and era classification for music browsing, automatic playlist generation and music recommendation, especially in electronic dance music.

## Session 4aNSa

## Noise and Engineering Acoustics: Jet and Rocket Noise I

Alan T. Wall, Cochair
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S. Hales Swift, Cochair
*Sandia National Laboratories, P.O. Box 5800, MS 1082, Albuquerque, NM 87123-1082*

**Chair's Introduction—9:30**

### *Contributed Papers*

**9:35**

**4aNSa1. Analysis of noise reduction methods for supersonic jets in the sideline and upstream directions using a statistical approach.** Trushant K. Patel (Mech. and Aerosp. Eng., Univ. of Florida, PO Box 116250, 939 Sweetwater Dr., Gainesville, FL 32611, trushant@ufl.edu) and Steven A. Miller (Mech. and Aerosp. Eng., Univ. of Florida, Gainesville, FL)

Jet noise remains a major source of concern for military personnel working aboard aircraft carriers. Broadband shock-associated noise (BBSAN) and fine-scale mixing noise (FSMN) are the main components of noise in the sideline and upstream directions. A decomposition approach based on the Navier–Stokes equations is used to identify the source of BBSAN and FSMN. Statistical models are developed to predict BBSAN and FSMN using the identified source terms. The prediction models use Reynolds-averaged Navier-Stokes solution as an argument. Comparisons of the noise radiated due to BBSAN and FSMN by three different nozzle types, i.e., a method of characteristics nozzle, a bi-conic nozzle, and a faceted nozzle are performed at different operating conditions. Different noise reduction mechanisms such as fluidic injection and corrugations on the nozzle wall are implemented on the faceted nozzle. We examine the change in source statistics and its correlation on the intensity and directivity of the radiated noise for BBSAN and FSMN. Noise reduction predictions relative to the baseline cases are compared for each nozzle modification. Finally, we analyze the noise source locations at different Strouhal number for both components of jet noise in the sideline direction.

**9:55**

**4aNSa2. Nonlinear modeling of tactical jet aircraft flyover noise.** Jacob A. Ward (Phys. and Astronomy, Brigham Young Univ., 50 S 500 W, Apt. 507, Salt Lake City, UT 84101, jacob.ward@live.com), Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Shane V. Lympany, and J. M. Downing (Blue Ridge Res. and Consulting, Asheville, NC)

Acoustic modeling of high-performance military aircraft is used to determine noise impacts and plan training operations. However, modeling accuracy to date has been limited because of nonlinear noise propagation, resulting in shock formation and greater high-frequency spectral levels. While prior work has investigated nonlinearity in noise propagation from static aircraft, this paper investigates the use of a nonlinear propagation algorithm based on the generalized Burgers equation to model noise propagation from F-35 flyovers. Waveforms recorded below the aircraft during a low-altitude flyover are used as the algorithm input and are propagated hundreds of meters. The sound pressure levels and spectra from the resulting waveforms are then compared with measured levels at the same distances. Using nonlinear modeling to predict the sound pressure levels significantly increases broadband spectral accuracy.

**10:15**

**4aNSa3. Far-field noise radiation characteristics of an afterburning military jet aircraft.** Michael S. Bassett (Dept. of Phys. and Astronomy, Brigham Young Univ., 562 N 200 E Apt 15, Provo, UT 84606, MichaelScottBassett@physics.byu.edu), Reese D. Rasband, Daniel J. Novakovich, Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Steven C. Campbell (Air Force Res. Lab, Beavercreek, OH), Frank S. Mobley, and Alan T. Wall (Air Force Res. Lab, WPAFB, OH)

Because of the potential noise impacts of tactical jet aircraft, detailed far-field measurements are valuable in developing and validating models. This paper describes far-field measurements taken of the T-7A Red Hawk trainer aircraft. Data were taken at five microphone arcs, over six engine runups from idle to afterburner. The arcs were of varying resolution, but generally spanned 30 to 160 deg (relative to the jet inlet) at distances from 19 to 230 m. The spectra, sound pressure level, and skewness of the time derivative of the pressure waveform are analyzed according to microphone position and engine condition. Measurement consistency over the different runups and across all conditions validates the dataset for future analysis. [Work supported by AFRL.]

**10:35**

**4aNSa4. Estimating the noise floor of sonic boom metrics across the USA.** William Doebler (NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, william.j.doebler@nasa.gov)

NASA is building the X-59 Quiet Supersonic Technology aircraft to produce low noise sonic booms for a series of community noise surveys across the USA. Survey participants will rate their perception of the low booms from supersonic X-59 flyovers. Several noise metrics are proposed to quantify the noise dose: A-, B-, D-, E-weighted Sound Exposure Level, Stevens Perceived Level, and Indoor Sonic Boom Annoyance Predictor. Sparse measurements across the survey area will be used to estimate community noise exposure. The level of these low booms may be comparable to the ambient noise level in some locations, leading to uncertainty in noise exposure estimations. This uncertainty may necessitate increased reliance on sonic boom propagation predictions for exposure estimation. Low-boom signal to ambient noise ratio is one way to quantify uncertainty in measured sonic boom levels. An empirical relationship between A-weighted ambient level and sonic boom metric levels is used in conjunction with the National Park Service's L50 SPL map to estimate sonic boom metric ambient levels across the USA. The estimate of ambient sonic boom metric levels will aid in X-59 test planning and execution.

## Session 4aNSb

# Noise: Noise Standards—Applications, Measurement Methods, Analyses, and Data Processing Involving ANSI/ASA S12 and/or ISO Noise Standards I

James E. Phillips, Cochair
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Christopher J. Struck, Cochair
*CJS Labs., 57 States Street, San Francisco, CA 94114-1401*

**Chair's Introduction—9:30**

### *Invited Papers*

**9:35**

**4aNSb1. A discussion on impulse noise exposure models in regard to small caliber firearm operation.** Steven C. Campbell (Ball Aerosp., 2875 Presidential Dr., Ste. 180, Beavercreek, OH 45324, sccampbe@ball.com), Alan T. Wall (711 Human Performance Wing, U.S. Air Force Res. Lab., WPAFB, OH), Reese D. Rasband (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), William J. Murphy (Div. of Field Studies and Eng., Noise and Bioacoustics Team, National Inst. for Occupational Safety and Health, Cincinnati, OH), and Frank S. Mobley (711 Human Performance Wing, U.S. Air Force Res. Lab., Wright-Patterson AFB, OH)

Impulse noise exposure is extremely prevalent in many military mission environments and, as such, it is critical that accurate hearing damage risk criteria be implemented. In 2017, the Air Force Research Laboratory (AFRL) conducted measurements at Quantico, Virginia on an outdoor shooting range in order to model exposure due to small firearm impulse noise. In this study, microphones were deployed across multiple arrays around an M16 rifle. Data were collected on the shooter for a selection of operational variables including shooter handedness, shooter height, and empty/occupied adjacent shooting lanes. A source model has been created through interpolation of data collected along a densely populated circular array surrounding the shooter at a radius of roughly three meters. By interpolating the array to achieve a fine angular resolution, the source model was then propagated via spherical spreading to generate a series of field point sound pressure levels. Via direct comparison, the simulated field points were validated against data collected from microphones outside of the source description array. Overall, this discussion provides a synopsis of previous work, future directions for research, and a discussion of the need for a unified, efficient exposure model for impulse noise. [Work supported by ONR.]

**9:55**

**4aNSb2. Acoustic standards for high-level impulse noise.** William J. Murphy (Div. of Field Studies and Eng., Noise and Bioacoustics Team, 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), Elizabeth B. Brokaw (The MITRE Corp., McLean, VA), and Raj K. Gupta (U.S. Army Medical Res. and Development Command, Dept of Defense Blast Injury Res. Coordinating Office, Fort Detrick, MD)

High-level impulse noise has been a significant source of hearing loss among active duty military personnel, veterans, law enforcement and recreational firearm users. The development of standards for estimating the risk of hearing loss due to such exposures is complicated by a lack of human exposure data. Following World War II, the Army Research Laboratory conducted a limited series of studies that led to the development of MIL-STD 1474, which was based on peak exposure level and envelope duration. In the late 1980s, the US Army recognized that large caliber weapons had different outcomes than small-caliber firearms. The Army conducted the Albuquerque Blast Overpressure Walk-up studies that led to extensive debate about how to estimate the risk of hearing loss from these sounds. The DoD Blast Injury Prevention Standards Recommendation Process evaluated several candidate noise exposure metrics including equivalent energy exposure, Auditory Hazard Assessment Algorithm for Humans, Integrated Cochlear Energy model, and Auditory Model 4.5. Weapon noise measurements were used to highlight how maximum permissible exposures provide a framework that relates the disparate predictions from the models.

**10:15**

**4aNSb3. A new noise exposure criteria is needed for complex noise.** Wei Qiu (Auditory Res. Lab., State Univ. of New York at Plattsburgh, 101 Broad St., Plattsburgh, NY 12901, qiuw@plattsburgh.edu), William J. Murphy (Div. of Field Studies and Eng., Noise and Bioacoustics Team, National Inst. for Occupational Safety and Health, Cincinnati, OH), and Mebian Zhang (Inst. of Environ. and Occupational Health, Zhejiang Provincial Centers for Disease Control and Prevention, Hangzhou, Zhejiang, China)

From the earliest standards for occupational noise exposure, impulse noise has been recognized as presenting an increased hazard for developing hearing loss. Animal research has focused primarily on the chinchilla as a surrogate for the human ear due to the similarity of sensitivity and frequency range of hearing. Research from the SUNY-Plattsburgh has demonstrated that hearing loss in chinchilla and

the kurtosis of the noise exposure are monotonically related for a given equivalent noise exposure level, $L_{Aeq}$. The risk of hearing loss increased with higher kurtosis to a point where the amount noise-induced hearing loss plateaued. SUNY-Plattsburgh, NIOSH, and the Chinese, Zhejiang Provincial Centers for Disease Control researchers have conducted a series of investigations with Chinese workers who have a history of noise exposure in a stable occupational setting. The findings from these investigations confirm the animal research, the risk of occupational hearing loss increases with an increased kurtosis of the noise exposure. This paper will examine the potential for a new noise exposure standard that incorporates both level and kurtosis.

### *Contributed Paper*

**10:35**

**4aNSb4. Performance of a ground-based microphone system for outdoor sound measurements.** Mark C. Anderson (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT 84606, anderson.mark.az@gmail.com), Kent L. Gee, Daniel J. Novakovich, Logan T. Mathews, and Zachary T. Jones (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

This paper discusses analyses of a ground-based outdoor microphone system designed to be weather-robust. This system employs an inverted half-inch microphone placed one-quarter inch above a thin plastic convex circular plate and enclosed in a dome windscreen. The current iteration of the "compact outdoor unit for ground-based acoustical recordings," nicknamed COUGARxt, is an improvement over its predecessor because of its thicker windscreen and its thinner plate made of a harder plastic material. One system characterization is anechoic chamber testing, where a sound source was placed at different elevation and azimuthal angles relative to the COUGARxt system to understand performance differences. Acoustical effects of plate orientation and the thicker windscreen are discussed. Another analysis consists of outdoor measurements in a windy but otherwise quiet environment. The COUGARxt system shows improved wind-noise rejection between 3 and 100 Hz, which could be important for improved detection of infrasound sources, including sonic booms and long-range launch vehicle noise.

THURSDAY MORNING, 10 DECEMBER 2020 — 11:15 A.M. TO 12:25 P.M.

## Session 4aNSc

## Noise and Engineering Acoustics: Jet & Rocket Noise II

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S. Hales Swift, Cochair
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**Chair's Introduction—11:15**

### *Contributed Papers*

**11:20**

**4aNSc1. Similarity spectral analysis of installed high-performance jet engine noise.** Kristi A. Epps (Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, kristi.epps5@gmail.com), Aaron B. Vaughn, Kevin M. Leete, Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Alan T. Wall (Air Force Res. Lab., WPAFB, OH)

Because the noise source mechanisms and radiation properties associated with high-thrust, tactical jet engines are not fully understood, full-scale measurements and analysis can shed significant insight. One method for examining spectral data is to compare them to empirical models for jet noise spectra. This paper compares measured near-field spectra from an installed F-404 engine with analytical similarity spectra for fine-scale mixing noise, large-scale mixing noise, and broadband-shock associated noise. This similarity spectral analysis enables us to determine spatial trends in overall level and peak frequency, and the relative importance of each type of noise radiation per location. This approach can be used to gain insights for different engine conditions as well as quickly compare to other aircraft.

**11:40**

**4aNSc2. Analysis of overall noise levels from space vehicle launches.** Logan T. Mathews (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, lmathew3@byu.edu), Kent L. Gee, and Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

A previous paper [L. T. Mathews *et al.*, *J. Acoust. Soc. Am.* **146**, 3043 (2019)] described acoustical measurements made of Falcon 9 space vehicle launches, as measured from various locations within a community near Vandenberg Air Force Base. This paper investigates the directivity pattern measured at one of the launches and compares it with historical and recent literature. Comparisons are made for peak directivity angle and corresponding level at that angle, directivity pattern shape, and calculated overall power level. Results are connected to relatively simple models based on a few engine parameters.

12:00

**4aNSc3. Comparative analyses far-field noise from GEM-63 solid rocket motor firings.** Jacob R. Smith (Phys. and Astronomy, Brigham Young Univ., 41 E 400 N Apt. 5, Provo, UT 84606, Jacobsmith1031@gmail.com), Clark O. Miller, Michael S. Bassett, Reese D. Rasband, Mark C. Anderson, Logan T. Mathews, and Kent L. Gee (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Prior far-field noise measurements from static rocket firings [e.g., see B. O. Reichman *et al.*, *Proc. Mtgs. Acoust.* **25**, 045006 (2017)] in northern Utah have yielded valuable physical insights. This paper describes a comparative analysis of measurements made at three GEM-63 solid-fuel booster firings. Measurements were made at angles between 40 and 120 deg (relative to the plume direction) and distances of 985–1832 nozzle diameters. This paper discusses the three measurement setups, as well as other notable features relevant for data analysis, such as terrain and local meteorology. Spectral and statistical analyses are used to understand the frequency and temporal characteristics of the noise as a function of angle for all three firings.

THURSDAY MORNING, 10 DECEMBER 2020 — 11:15 A.M. TO 12:45 P.M.

## Session 4aNSd

## Noise, Architectural Acoustics, and Signal Processing in Acoustics: Noise Standards—Applications, Measurement Methods, Analyses, and Data Processing Involving ANSI/ASA S12 and/or ISO Noise Standards II

James E. Phillips, Cochair
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Christopher J. Struck, Cochair
*CJS Labs., 57 States Street, San Francisco, CA 94114-1401*

**Chair's Introduction—11:15**

### *Invited Papers*

11:20

**4aNSd1. An update to ANSI/ASA S12.2-2019: Criteria For evaluating room noise—Where to go next?** Nicholas Sylvestre-Williams (Aercoustics Eng. Ltd., 165-50 Ronson Dr., Toronto, ON M9W 1B3, Canada, nicholass@aercoustics.com)

The S12.2 standard provides methods for evaluating room noise such as NC curves or overall dBA levels. It has been around, in a rather similar form, for more than 25 years. While some concepts have moved from the main section to the annex (e.g., NCB), other concepts remain unchanged. A survey was taken in 2019 to evaluate the use and functionality of the various elements of the standard (e.g., the degree of usage of NC versus dBA vs NCB versus PNC vs RNC versus RCMkII). Additional items for consideration into the a revision of the standard were polled (e.g., 1/3 Octaves, minimum background levels, etc.) This session will give a brief overview of the standard and significant time will be spent on the next steps and ideas for changes to the standard. Feedback will be welcome and additional time will be given for group discussion.

11:40

**4aNSd2. The evolution of standard S12.60 acoustical performance criteria, design requirements, and guidelines for schools.** Stephen J. Lind (LindAcoustics LLC, 1108 Valley Vue Dr., Onalaska, WI 54650, stephen.j.lind.ut88@gmail.com)

The committee on standards for the Acoustical Society of America was asked by the US Access board and others to develop standards for acceptable acoustics in classrooms. The effort resulted in a standard comprised of several parts. The first document was approved in 2002 and this later became part 1. The initial desire was to have a federal rulemaking and incorporate the standard into coverage for the Americans with Disabilities Act. Part 2 was approved in 2009 and provided guidance for relocatable learning spaces. The intent to have a federal rulemaking was thwarted by political factors so Part 1 was revised in 2010 with the intent to put the document into a format that would allow its incorporation or use in building codes. Part 3 is still being developed, but it is intended for Information Technology Equipment in Classrooms. Part 4 was approved in 2019 and gives guidance for education spaces such as gymnasia and other spaces used for physical education.

12:00

**4aNSd3. Education and healthcare: A look at ASA/ANSI S12.60 and ASA/ANSI S12.70.** Kenneth W. Good (Armstrong World Industries, Inc., 2500 Columbia Ave., Lancaster, PA 17601, kwgoodjr@armstrong.com)

Two of the most important spaces for society are where we learn and where we heal. This session will take a brief review of the ASA/ANSI S12.60 standards for education spaces and ASA/ANSI S12.70 speech privacy in healthcare standard. The discussion also will include architectural trends and what updates the respective working groups are addressing.

12:20

**4aNSd4. The reinstatement of Working Group 41 (ANSI S12.66) guidelines for developing a community noise ordinance or regulation: First steps.** David s. Woolworth (Roland, Woolworth & Assoc., 356 County Rd. 102, Oxford, MS 38655-8604, dwoolworth@rwaconsultants.net)

Working Group 41 of the ASA standards committee oversaw the development of ANSI-ASA S12.66 Guidelines for Developing a Community Noise Ordinance or Regulation, and has become active again after becoming dormant in 2013. The original goal of the working group was to create a model ordinance that could be tailored to a particular community by the local authorities, however over the 13+ years of the development effort it was evident that a "model" ordinance takes on the characteristics of a holy grail. A more holistic approach was adopted to examine community needs and resources, and considerations for judicial, executive, and technical limitations. Furthermore, the evolution of soundscape methods provide new tools for improving a community's involvement in the assessment of its needs. This paper will provide an overview of the working group's activity since Spring 2020.

## Session 4aPAa

## Physical Acoustics, Engineering Acoustics, Structural Acoustics and Vibration, and Biomedical Acoustics: Acoustofluidics I

Charles Thompson, Cochair
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John M. Meacham, Cochair
*Mechanical Engineering & Materials Science, Washington University in Saint Louis, 1 Brookings Dr., Jubel Hall, Rm 203K, Saint Louis, MO 63130*

Kedar Chitale, Cochair
*Flodesign Sonics, Inc., 380 Main Street, Wilbraham, MA 01095*

Max Denis, Cochair
*University of the District of Columbia, 4200 Connecticut Ave. NW, Washington, DC 20008*

**Chair's Introduction—9:30**

### *Invited Papers*

**9:35**

**4aPAa1. Acoustic field design in microfluidic geometries via Huygens-Fresnel diffraction and deep neural networks.** David Collins (Univ. of Melbourne, 203 Bouverie St., Rm. 103, Victoria 3010, Australia, david.collins@unimelb.edu.au), Sam Raymond (Stanford, Cambridge, MA), Ye Ai (Singapore Univ. of Technol. and Design, Singapore, Singapore), John Willams (Massachusetts Inst. of Technol., Cambridge, MA), Richard O'Rorke, and Mahnoush Tayebi (Singapore Univ. of Technol. and Design, Singapore, Singapore)

Creating defined nonuniform acoustic fields, with particle capture locations beyond the grids and lines that are typically generated, has a range of applications in microfluidic systems. In this work, we use a unique interaction between polydimethylsiloxane (PDMS) channel geometries and a travelling substrate wave to explore the creation of nonuniform acoustic fields for microscale pattering. Surface acoustic waves (SAWs) are ideal for generating high-frequency, microscale wavelengths, though conventionally generate particle pattering via the interference between opposing SAW transducers, creating a standing SAW. Our recent work, however, has shown that patterning can alternatively be created using only a single travelling SAW and channel boundaries via diffractive effects, where fringe spacing is a function of the channel wall orientation. Unique to SAW, this creates a condition where the channel shape directly impacts the acoustic field that results, and therefore the possibility of designing acoustic fields by defining the channel geometry. To solve this inverse problem, we implement a machine-learning approach based on a Deep Neural Networks (DNN) that can define channel geometries resulting in desired acoustic fields. This submitted work will introduce the mechanisms via which this diffractive patterning occurs and the implementation of the DNN design process.

**9:55**

**4aPAa2. Experimental assessment and modeling of acoustophoretic particle trajectories.** Rune Barnkob (Heinz-Nixdorf-Chair of Biomedical Electronics, Tech. Univ. of Munich, Einsteinstraße 25 (Geb. 522), At TranslaTUM, Campus Klinikum rechts der Isar, Munich, Bavaria 81675, Germany, rune.barnkob@tum.de)

The field of acoustofluidics, the employment of acoustic forces in microfluidic systems, is receiving growing attention for use in biomedical applications due to its use for label-free and non-invasive separation and manipulation of cells and other biological particles. The field is young but has over the last decade undergone an important development towards clinical and industrial application, in particular due to an improved understanding of the underlying fundamental physical aspects. Here, theoretical modeling and experimental assessment of acoustophoretic particle trajectories have played an essential role, e.g., for the measurement of the *in situ* pressure amplitudes and resonance quality factors or for the understanding of the critical particle size for which the acoustophoretic particle motion is dominated by the acoustic streaming drag or the primary acoustic radiation force. These phenomena are typically of complex three-dimensional character and can only be fully understood through three-dimensional quantification such as through three-dimensional tracking of the acoustophoretic particle motion, e.g., via the General Defocusing Particle Tracking method. The importance of three-dimensional particle tracking will be demonstrated by application to the acoustophoretic particle motion in soft-walled polymer chips driven by surface acoustic waves as well as in hard-walled square glass capillaries driven by bulk piezo transducers.

**10:15**

**4aPAa3. Interfacial tension and acoustofluidics: some connections, instabilities, experimental results, and theory.** Philip L. Marston (Phys. & Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

It is timely to examine early acoustofluidic studies involving surface tension phenomena. The dynamics of fluid cylinders and the Plateau-Maxwell-Rayleigh interfacial instability have been reviewed [P. L. Marston, *Phys. Fluids* **29**, 029101 (2017)]. Here, some related acoustofluidic experiments on liquid cylinders are noted [S. F. Morse *et al*., *Phys. Fluids* **8**, 3 (1996); M. J. Marr-Lyon *et al*., *Phys. Rev. Lett.* **86**, 2293 (2001)]. Those were after modulated radiation pressure investigations of the shape dynamics of levitated drops and bubbles [P. L. Marston, *J. Quant. Spectrosc. Radiat. Transfer* **254**, 107226 (2020)]. Especially noteworthy are the subtleties of acoustically trapping and deforming bubbles in water larger than the size for monopole resonance [T. J. Asaki, P. L. Marston, and E. H. Trinh, *J. Acoust. Soc. Am.* **93**, 706 (1993)]. That ability facilitated a series of measurements of bubble interfacial dynamics and shape [Asaki *et al*., *J. Acoust. Soc. Am.* (1994-1997); *J. Fluid Mech.* (1995); *Phys. Rev. Lett.* (1995)]. A related discussion of the quadrupole projection of the radiation stress is noteworthy [P. L. Marston, *Phys. Rev. E* 100, 057001 (2019)] along with early publications reporting the break-up of drops and bubbles in water through the application of modulated acoustic radiation forces.

**10:35**

**4aPAa4. Acoustofluidic equations governing fluid-elastic media with suspended particles and shear elastic properties.** 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 (Cape Cod Inst. for Sci. and Eng., Troy, NY), and Elisabeth M. Brown (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

Understanding physics of complex media is generally helped by one's having a complete set of partial differential equations that govern the motion and interactions within the media. A single scalar wave equation is usually not sufficient for understanding all the acoustic properties of interest. An example of such a set of equations are those developed by M. A. Biot (1956) for low frequency propagation of compressional, shear, and other waves in poroelastic media. An analogous set of equations is here developed for a medium that has three primary phases—(1) a fluid, (2) a gossamer array of thin platelets held together by van-der-Waals forces, and (3) a disperse and somewhat random array of larger solid particles held in suspension by the gossamer array. Marine sediment mud is one example of such a medium. Because of the gossamer array. the medium resists static shear, and its compressibility is primarily caused by the fluid. Using various physical principles, one can formulate different elastical-mechanical equations for each of the three phases, or even for subcategories of the phases (such as for suspended particles within different size ranges). The phases interact via different physical mechanisms, such as viscous forces exerted by the fluid on the suspended particles. Examples are given for the use of these equations and of their implications. [This material is based upon research supported by, or in part by, the U. S. Office of Naval Research under award numbers N00014-15-2039, N00014-18-1-2439, and N00014-19-1-2636.]

# Session 4aPAb

## Physical Acoustics, Engineering Acoustics, Structural Acoustics and Vibration, and Biomedical Acoustics: Acoustofluidics II

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*Mechanical Engineering & Materials Science, Washington University in Saint Louis, 1 Brookings Dr., Jubel Hall, Rm. 203K, Saint Louis, MO 63130*

Kedar Chitale, Cochair
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Max Denis, Cochair
*University of the District of Columbia, 4200 Connecticut Ave. NW, Washington, D.C. 20008*

**Chair's Introduction—11:15**

### *Invited Paper*

**11:20**

**4aPAb1. Airflow-acoustic interactions in speech production research.** Christine H. Shadle (Haskins Labs., 300 George St. Ste. 900, New Haven, CT 06511, shadle@haskins.yale.edu)

Speech production involves air traveling from a reservoir (the lungs) through various ducts (the bronchi and trachea, the vocal and nasal tracts), past various structures at which sounds can be generated by the flow of air. Some structures (e.g., the vocal folds or the tongue tip) can be positioned so as to self-oscillate, generating quasi-periodic sound; some (e.g., the lips or the tongue tip) can be positioned to create a constriction small enough to generate turbulence, and thus turbulence noise. These sounds excite the resonances of the tract. They can also interact, as when the sound generated by vocal fold vibration modulates the turbulent jet formed at a downstream constriction. It would seem to be straightforward to predict the sound that propagates through the tract and is radiated externally, except that the tract changes shape continually; the walls of the tract vary in compliance, from hard (e.g., teeth) to yielding (e.g., cheeks); and in spite of co-opting many types of medical imaging, it is very difficult to measure the interior shape of the tract accurately and without disturbing either the speech or the speaker. Some acousto-fluidic interactions that occur during speech, and the ways they have been modeled, are discussed.

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### *Contributed Papers*

**11:40**

**4aPAb2. Direct measurement of small spherical particle rotation driven by the acoustic viscous torque.** Christoph Goering (D-MAVT, ETH Zürich, Tannenstr. 3, ETH - CLA H27, Zürich 8092, Switzerland, goering@imes.mavt.ethz.ch), Andreas Lamprecht (D-MAVT, ETH Zürich, Zürich, Switzerland), Iwan A. Schaap (None, Oldenburg, Germany), and Jürg Dual (D-MAVT, ETH Zürich, Zurich, Switzerland)

We report a new method for measuring fast rotations up to 13500 rpm of micrometer sized spherical particles utilizing an optical trap [A. Lamprecht *et al.*, *Lab Chip* **16**, 2682–2693 (2016)]. In an acoustofluidic flow cell, a single spherical particle rotates due to the acoustic viscous torque produced by two phase-shifted orthogonal standing waves. The Lorentzian power spectrum of the trapped and rotating particle has additional peaks at frequencies that correlate with its rotational speed. In one of our experiments the thickness of the viscous boundary layer $\delta$ around the particle is roughly as large as the particle radius R itself. We use a water glycerol mixture with a dynamic viscosity of $\mu\_f_f = 0.06$ Pa s to increase $\delta$. Our experiment validates recent numerical research that predicts power rotational speeds for the case $\delta \approx$ R compared to the simplified theoretical formula valid for $\delta \ll$ R [Lee and Wang, *J. Acoust. Soc. Am.* **85**, 1081–1088 (1989)].

**12:00**

**4aPAb3. Femtoliter acoustofluidics.** Naiqing Zhang, Amihai Horesh (Mech. and Aerosp. Eng., Univ. of California San Diego, La Jolla, CA), Ofer Manor (Wolfson Faculty of Chemical Eng., Technion, Haifa, Israel), and James Friend (Mech. and Aerosp. Eng., Univ. of California San Diego, 9500 Gilman Dr. MC0411, MADLab SME344K, La Jolla, CA 92093, jfriend@ucsd.edu)

Working with extremely small fluid samples has been difficult due to evaporation and the dominance of surface and viscous forces. Recent work has demonstrated that manipulation of fluids and their contents at these scales is now possible using MHz-order acoustic waves, not only due to

acoustic streaming but other, newly discovered nonlinear interactions between the acoustic wave and the channel. Here, we show that MHz-order surface acoustic waves (SAW) can manipulate 10–100 femtoliter fluid droplets within fully transparent, high-aspect ratio (100 nm tall, 20–100 $\mu$m wide, 5 mm long) nanoslits fabricated via a direct, room temperature bonding method for lithium niobate. The droplets are entrapped by capillary forces in discrete positions, enabling droplet transport, splitting, merging, and mixing, all at length scales far below what has been reported in the literature to date. Enabled by the intense accelerations made possible from acoustic waves, we detail the new form of droplet propulsion and manipulation with a detailed theoretical model that captures the salient physics. Our results suggest that nanofluidics is essentially made practical via acoustics.

**12:20**

**4aPAb4. Spatial and frequency selective acoustically driven streaming and its application to boundary-induced pattern formation.** Charles Thompson (Elec. and Comput. Eng., UMASS Lowell, 1 Univ. Ave., Lowell, MA 01854, charles_thompson@uml.edu), Kavitha Chandra (Elec. and Comput. Eng, UMASS Lowell, Lowell, MA), and Eric Saadatmand (Elec. and Comput. Eng, UMASS Lowell, Lowell, NH)

The problem of spatial and frequency selective acoustic streaming induced in a bifurcated channel using an elastic membrane is examined. The acoustic wavelength is much longer than the channel dimensions. The time-averaged fluid motion can be controlled by adjusting the pressure gradient and the impedance of the membrane. The fluid's dynamic behavior is governed by the relative magnitude of three parameters, the oscillatory Reynolds, Strouhal, and streaming Reynolds numbers. The behavior in the large oscillatory Reynolds and Strouhal number limit is of particular interest. The solution is given by singular asymptotic series expressed in terms of the reciprocal of the Strouhal number and matched using the method of matched asymptotic expansions.

THURSDAY MORNING, 10 DECEMBER 2020 — 9:30 A.M. TO 10:15 A.M.

## Session 4aPPa

## Psychological and Physiological Acoustics: Clinical and Aging Populations (Poster Session)

Authors will be at their posters from 9:30 a.m. to 10;15 a.m.

### *Contributed Papers*

**4aPPa1. Relationship between environmental sound imagery and environmental sound identification in cochlear implant listeners.** Breanna L. Corle (Audiol., Rush Univ., 600 S Paulina St., Chicago, IL 60608, breanna_corle@rush.edu), Valeriy Shafiro (Audiol., Rush Univ., Chicago, IL), Kara J. Vasil, and Aaron C. Moberly (Otolaryngol., The Ohio State Univ., Columbus, OH)

Identification of common environmental sounds ("car honking," "baby crying," "waves") is a continuing challenge for many adult postlingual cochlear implant (CI) listeners. This study investigated whether environmental sound imagery, used to characterize the strength of underlying mental representations, relates to CI listeners' environmental sound identification. CI participants first read the names of 24 individual sounds and rated how familiar, pleasant, complex, and easy to imagine each sound was on a 5-point scale. Next, they heard and identified each sound. Rating responses were transformed based on how each sound's rating corresponded to its identification accuracy to obtain an imagery correspondence index (ICI). A higher ICI indicates a closer agreement between a sound's rating and its identification accuracy. Across participants results indicated moderate-to-high correlations between identification accuracy and ICI scores only for familiarity and ease of imagining but not complexity and pleasantness. A similar correlation pattern with raw ratings was observed across sounds but not across participants. These results suggest that aspects of environmental sound imagery can be predictive of auditory perception of real-world sounds in CI listeners and should be considered in future diagnostic and rehabilitation applications.

**4aPPa2. Effects of context in environmental sound perception in older normal hearing and cochlear implant listeners.** Katie I. Swail (Audiol., Rush Univ., 2724 W Logan Blvd., Apt. 3, Chicago, IL 60647, katie_swail@rush.edu), Valeriy Shafiro (Audiol., Rush Univ., Chicago, IL), Kara J. Vasil, Aaron C. Moberly (Otolaryngol., The Ohio State Univ., Columbus, OH), Jasper Oh, and Melissa Malinasky (Audiol., Rush Univ., Chicago, IL)

The use of context by individuals with hearing loss has been extensively studied in speech, but not in environmental sound perception. The effect of context on environmental sound perception was investigated in 26 older normal hearing (ONH) and 46 adult postlingual cochlear implant (CI) listeners. Participants first identified 24 individual environmental sounds in isolation and then were presented with the same sounds arranged in 3–5 sound sequences. Half of the sequences were either contextually coherent (i.e., likely to be heard together) and half were incoherent. Results revealed greater accuracy of ONH than CI listeners in identification of isolated environmental sounds. However, no group differences were observed in the identification of sounds sequences. In both groups contextually coherent sequences were identified more accurately than incoherent ones, while accuracy decreased with sequence length. Sequence length interacted with context, with context effect increasing for longer sequences. These findings demonstrate that environmental sound identification remains challenging for CI listeners. Nevertheless, both CI and ONH listeners utilize context to aid in environmental sound identification in perceptually challenging tasks associated with a greater working memory load. Future research may further assess the relationship between environmental sound perception and other ecologically significant aspects of electric hearing.

**4aPPa3. Vocal emotion identification and quality of life in adult cochlear implant users.** Stephanie Strong (Commun. Disord. and Sci., Rush Univ. Medical Ctr., 600 S Paulina St., Chicago, IL 60612, stephanie_g_strong@rush.edu), Aaron C. Moberly, Kara J. Vasil (Otolaryngol., The Ohio State Univ., Columbus, OH), and Valeriy Shafiro (Commun. Disord. and Sci., Rush Univ. Medical Ctr., Chicago, IL)

Self-report quality of life questionnaires, such as the Nijmegen Cochlear Implant Questionnaire (NCIQ), are increasingly paired with speech intelligibility tasks to assess cochlear implant (CI) outcomes. Previous research [Luo *et al.*, *J. Acoust. Soc. Am.* **144**, EL429–EL435 (2018)] found that vocal emotion identification was positively correlated with the six NCIQ subdomains. The current study attempted to replicate and extend previous findings, using similar procedures with a new sample of CI listeners. Eighteen postlingually implanted adults were randomly presented with 10 semantically neutral sentences spoken by one female talker expressing one of five emotions (happy, sad, angry, scared, and neutral). In addition to NCIQ, participants completed SSQ12 and HHIE. The overall emotion identification accuracy was 66% correct. No significant associations were found between emotion identification and any NCIQ or HHIE subscales. There were significant positive correlations between emotion identification and SSQ12 (spatial subscale and total score). The present NCIQ results diverged from the previous findings, suggesting at best a tenuous relationship between emotion identification quality-of-life measures. However, the reduced emotion identification scores of CI users suggest that these tasks may have clinical utility in CI assessment and rehabilitation.

**4aPPa4. Perception of vowels and consonants in cochlear implant users.** Melissa Malinasky (Rush Univ. Medical Ctr., Chicago, IL, melissa_malinasky@rush.edu), Valeriy Shafiro (Commun. Disord. and Sci., Rush Univ. Medical Ctr., Chicago, IL), Aaron C. Moberly, and Kara J. Vasil (Otolaryngol., The Ohio State Univ., Columbus, OH)

Although cochlear implants (CI) improve speech intelligibility, they introduce distortions of the sensory input to the auditory system, and modify the transmission of acoustic cues. This study evaluated perception of consonants and vowels in experienced postlingual adult CI users. Twenty-five experienced CI users were tested with The Modified Rhyme Test (MRT) of consonant perception (word-initial and word-final) and a closed-set vowel test containing 12 vowels in the /hVd/ context. Overall consonant perception accuracy was 80%. No significant difference was observed in identification of word-initial or word-final consonants. There was significant variation in consonant accuracy in terms of place and manner, but not voicing. For place, alveolar consonants had the lowest accuracy. For manner, nasals were the least identifiable. Overall vowel identification accuracy was 76%. Accuracy was higher for front vowels than back vowels. The three most accurately identified vowels were (/i/), (/I/), and (/ʊ/). The three least accurately identified vowels were (/a/), (/ɔ/), and (/ʊ/). There was a considerable variation in consonant and vowel perception across CI users. The pattern or consonant and vowel confusions can inform CI programming and rehabilitation to improve speech perception.

**4aPPa5. Enhancing the temporal fine structure with the temporal limits encoder for cochlear implants: Effects on pitch discrimination.** Huali Zhou (Acoust. Lab., School of Phys. and Optoelectronics, South China Univ. of Technol., Rm. 301, Bldg. 18, No.381, Wushan Rd., Tianhe District, Guangzhou 510641, China, mshualizhou@mail.scut.edu.cn), Guangzheng Yu, and Qinglin Meng (Acoust. Lab., School of Phys. and Optoelectronics, South China Univ. of Technol., Guangzhou, China)

To enhance the mostly discarded temporal fine structure in modern cochlear implant (CI) strategies, a temporal-limits-encoder (TLE) strategy was proposed by downshifting the high-frequency-band-limited signal to a low-frequency-temporal-pitch-limits range of CIs [Meng *et al.*, *J. Acoust. Soc. Am.*(2016)]. This study investigates pitch perception with a TLE strategy compared with a standard advanced combinational encoder (ACE) strategy. Seven CI subjects were tested in a complex-tone-pitch discrimination task measuring the fundamental frequency difference limens (F0DLs) at four reference F0s (250, 313, 1000, and 1063 Hz, which are the center and upper cross-over frequencies of two bands). Results show that (1) the CI listeners generally had lower F0DLs with TLE than with ACE (group mean F0DL benefits of TLE over ACE of 5.0, 9.6, 0.5 and 4.3 percentage points at the four reference F0s, respectively) and (2) the two strategies had comparable sentence recognition performance in both quiet and noisy conditions. These findings suggest that the slowly varying TFS introduced by TLE is feasible in pitch discrimination for CI listeners and is not significantly detrimental to sentence recognition. This discrimination advantage can be explained by larger differences in the temporal fluctuations on individual channels with TLE than with ACE.

**4aPPa6. Using cluster analysis and hearing histories to predict performance outcomes among bilateral cochlear-implant users.** Brittany N. Jaekel (Hearing and Speech Sci., Univ. of Maryland, 0221 LeFrak Hall, College Park, MD 20742, jaekel@umd.edu) and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland, College Park, MD)

Cochlear-implant researchers are often interested in how participants' hearing histories impact performance. For example, it is common to include duration of deafness, age at onset of deafness, and age at implantation as covariates in statistical analyses. Unfortunately, multi-collinearity often exists among hearing history variables, which can affect the convergence of statistical models. This problem could be alleviated by using cluster analysis, a statistical method that can sort participants into groups based on their similarity on a set of variables. Cluster membership, rather than collinear hearing history variables, can then be entered into statistical models. To measure the efficacy of this method in cochlear-implant research, we performed a cluster analysis on the bilateral cochlear-implant users in our lab's database with complete hearing history information ($n = 53$). Bilateral cochlear-implant users were chosen for this initial analysis to measure whether across-ear differences affect outcomes. Four clusters emerged based on demographic and hearing history variables. Preliminary analyses showed that cluster membership could serve as a useful predictor for performance outcomes in bilateral cochlear-implant users. Validating clusters in the greater cochlear-implant user population could result in useful information for audiologists counseling potential implant candidates about expected outcomes.

**4aPPa7. Launching the first "Clarity" Machine Learning Challenge to revolutionise hearing device processing.** Michael A. Akeroyd (School of Medicine, Univ. of Nottingham, Hearing Sci. Bldg., University Park, Nottingham NG7 2RD, United Kingdom, michael.akeroyd@nottingham.ac.uk), Jon P. Barker (Dept. of Comput. Sci., Univ. of Sheffield, Sheffield, United Kingdom), Trevor J. Cox (Acoust. Res. Group, Univ. of Salford, Salford, Greater Manchester, United Kingdom), John Culling (School of Psych., Univ. of Cardiff, Cardiff, United Kingdom), Simone Graetzer (Acoust. Res. Group, Univ. of Salford, Salford, United Kingdom), Graham Naylor, Eszter Porter (School of Medicine, Univ. of Nottingham, Nottingham, United Kingdom), and Rhoddy Viveros Muñoz (School of Psych., Univ. of Cardiff, Cardiff, United Kingdom)

Recent advances in machine learning raise the prospect of radically improving how hearing devices deal with speech in noise and so improve many aspects of health and well-being for an aging population. In many other aspects of speech processing, rapid transformations have been enabled by a research tradition of "open challenges." Thus, to catalyse such progress for hearing devices we launch this month (Dec 2020) a new machine-learning challenge series, "Clarity" (claritychallenge.org) The first challenge will focus on sentences from single talkers in a moderately reverberant environment with typical background noises for the home. We will supply all these recordings (including a new multi-talker database of speech sentences) together with generative models to combine them in sufficient quantities for machine learning. We will also supply a comprehensive set of "baseline," modular models of hearing impairment, hearing aids, speech intelligibility and speech quality. The tasks for challenge entrants is then to build on these, developing new and improved approaches for hearing device signal processing so that the intelligibility of these sentences is maximised for hearing-impaired listeners.

**4aPPa8. Cross-frequency weights for loudness, pitch, and duration and their relation to speech recognition in normal-hearing and hearing-impaired listeners.** Elin Roverud (Boston Univ., 635 Commonwealth Ave., Dept. of Speech, Lang. & Hearing Sci., Boston, MA 02215, erover@bu.edu), Judy R. Dubno (Medical Univ. of South Carolina, Charleston, SC), and Gerald Kidd (Boston Univ., Boston, MA)

Many hearing-impaired (HI) listeners have uneven hearing loss across frequency, which could alter their weighting of cross-frequency information as compared to normal-hearing (NH) listeners. This has been suggested in previous psychophysical studies where listeners made loudness/level judgments of a multitone complex in which the level of each component was randomly varied trial to trial. The frequency components for which level variations correlated most strongly with subject responses indicate those given the most weight. However, the extent to which those weights are similar to weights for perception of pitch and duration—and how weights relate to the use of spectral information in speech—is not known. Here, the same NH and HI listeners performed weighting tasks involving the perception of loudness, pitch and duration. Recognition of low-pass filtered speech as a function of cutoff frequency was also measured. We tested the hypothesis that HI listeners would consistently downweight/underutilize frequency regions with more severe hearing loss, and that this would be apparent for weights across multiple stimulus dimensions and in speech recognition changes with bandwidth. Relationships among weights for NH and HI listeners for three stimulus features and bandwidth-related changes in speech recognition will be discussed. [Work supported by NIH NIDCD K01DC016627.]

**4aPPa9. Aging-related perceptual-grouping and attention-switching deficits in frequency discrimination amid task-irrelevant stimulus variability.** Blas Espinoza-Varas (Commun. Sci. & Disord., OU Health Sci. Ctr., 1200 N. Stonewall Ave., Oklahoma City, OK 73117-1215, blas-espinoza-varas@ouhsc.edu)

In elderly adults, failure to regroup auditory events and reallocate attention can defeat ignoring conflicting, task-irrelevant (CTI) information; specifically, ignoring CTI duration ($\Delta$Tc) or level ($\Delta$Lc) differences to focus on task-relevant frequency differences ($\Delta$Fr). Frequency Discrimination Thresholds (FDTs) were measured in 3I2AFC paradigms presenting a standard (Ts) followed by two comparisons ($T_1$, $T_2$); each interval presented tones either in isolation or trailed by captor tones. In young (YA) and in older (OA) adults, FDTs were measured without conflicting differences and amid $\Delta$Tc or $\Delta$Lc presented simultaneously with $\Delta$Fr or sequentially in the captor; FDT elevations relative to "without-conflict" FDTs indicated difficulty ignoring CTI differences. Because $\Delta$Lc disrupted the without-conflict discrimination strategy, amid $\Delta$Lc (but not amid $\Delta$Tc) discerning $\Delta$Fr required switching attention and regrouping auditory events: segregating Ts from $T_1$, $T_2$ with targets in isolation, or captors from targets in sequential conditions. Higher FDTs in OA than in YA obtained only if regrouping auditory events was mandatory. Facilitating segregation with increasing target-captor separation improved the FDTs of OA in all CTI difference conditions, but only amid $\Delta$Lc in YA. Regrouping auditory events and reallocating attention entailed shifting from bottom-up, automatic to top-down, schema-driven sequential grouping.

**4aPPa10. The influence of target/masker fundamental frequency contour similarity on masked-speech recognition for older and younger adults.** Peter A. Wasiuk (Dept. of Psychol. Sci., Case Western Reserve Univ., 10730 Euclid Ave., Apartment 1204, Cleveland, OH 44106, paw70@case.edu), Mathieu Lavandier (Laboratoire Génie Civil et Bâtiment, Univ Lyon, ENTPE, Lyon, France), Emily Buss (Dept. of Otolaryngology/Head and Neck Surgery, Univ. of North Carolina, Chapel Hill, Chapel Hill, NC), Jacob Oleson (Dept. of Biostatistics, Univ. of Iowa, Iowa City, IA), and Lauren Calandruccio (Dept. of Psychol. Sci., Case Western Reserve Univ., Cleveland, OH)

Older adults with sensorineural hearing loss (SNHL) have greater difficulty recognizing target speech in multi-talker backgrounds than young adults with normal hearing (NH). These deficits are due in part to a decreased ability to utilize perceptual differences between target and masker speech to segregate sound sources. For young NH listeners, fundamental frequency (f0) contour (i.e., dynamic f0 movement) has been demonstrated to be an effective stream segregation cue when acoustic-perceptual differences in target and masker f0 contours exist. This project examined if older adults with SNHL can utilize f0 contour differences between target and masker speech to improve target speech recognition. F0 contours of target and masker speech were systematically manipulated to be either *flat, normal,* or *exaggerated*. Computational modeling was conducted to estimate differences in energetic masking across listening conditions. Effects of age and hearing loss were differentiated in the statistical model. Results indicated that older adults with SNHL were unable to take advantage of target/masker f0 contour differences to improve target speech recognition in the same manner as young adults, even with prescriptive amplification applied to the speech stimuli. This may contribute to difficulties experienced by older adults in real-world multi-talker environments, even when using amplification technologies.

**4aPPa11. Autism, a hearing problem?** Robert H. Cameron (Eng. Technol., NMSU (retired), 714 Winter Dr., El Paso, TX 79902-2129, rcameron@elp.rr.com)

At the San Diego Acoustical Society meeting, on my poster, was a picture of an eight year old child with a headset that allowed him to hear only with his dominant right ear. He was focused on his homework. Days before he was running down the street into traffic, so frustrated by people repeatedly saying pizza and not understanding. He repeated what he heard, "Sissa." I posit that different frequencies of the signal from each ear propagate at different speeds in the central nervous system of an autistic child so that the waveform is distorted. To demonstrate this I propose to present this child with a headset with sine, and more complex waves, of different frequencies within the speech range at his right ear and the same signal 180 deg out of phase at his left ear, and ask him to point to where the sound is coming from. (A previous experiment with a "non-autistic" child indicated that he would sense that the sound comes from the center of his head.) Where he points to should indicate whether or not all frequencies travel at the same speed. In addition, I will have selected from the list of the most common English words, words I believe he has heard as well as words he is unlikely to have heard and ask him to repeat them, first with hearing blocked in less dominate ear then unblocked to see how long it takes to correct the mispronunciation in each case.

# Session 4aPPb

## Psychological and Physiological Acoustics and Speech Communication: Remote Testing for Auditory and Speech Research I (Poster Session)

Authors will be at their posters from 10:15 a.m. to 11:00 a.m.

### *Contributed Papers*

**4aPPb1. Remote testing for psychological and physiological acoustics: Initial report of the ASA P&P Task Force on Remote Testing.** G. Christopher Stecker (Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68131, cstecker@spatialhearing.org), Jordan A. Beim (Univ. of Minnesota, Minneapolis, MN), Hari Bharadwaj (Speech, Lang., and Hearing Sci., Purdue Univ., West Lafayette, IN), Adam K. Bosen (Boys Town National Res. Hospital, Omaha, NE), Emily Buss, Meredith Braza (Otolaryngol./Head and Neck Surgery, Univ. of North Carolina School of Medicine, Chapel Hill, NC), Anna C. Diedesch (Commun. Sci. & Disord., Western Washington Univ., Bellingham, WA), Claire M. Dorey (Univ. of Florida, Gainesville, FL), Andrew R. Dykstra (Biomedical Eng., Univ. of Miami, London, ON, Canada), Richard Freyman (Commun. Disord., Univ. of Massachusetts, Amherst, MA), Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR), Raymond L. Goldsworthy (Caruso Dept. of Otolaryngology-Head and Neck Surgery, Univ. of Southern California, Keck School of Medicine, Los Angeles, CA), Lincoln Gray (Commun. Sci. and Disord., James Madison Univ., Harrisonburg, VA), Eric C. Hoover (Dept. of Hearing and Speech Sci., Univ. of Maryland, Columbia, MD), Antje Ihlefeld (Dept. of Biomedical Eng., New Jersey Inst. of Technol., Newark, NJ), Thomas Koelewijn (Otorhinolaryngology, Univ. Medical Ctr. Groningen, Groningen, The Netherlands), J. G. Kopun (Boys Town National Res. Hospital, Omaha, NE), Juraj Mesik (Univ. of Minnesota, Minneapolis, MN), Ellen Peng (Univ. of Wisconsin-Madison, Madison, WI), Virginia M. Richards (Dept. of Cognit. Sci., Univ. of California, Irvine, CA), Yi Shen (Dept. of Speech and Hearing Sci., Univ. of Washington, Seattle, WA), Daniel E. Shub (Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., Bethesda, MD), Jonathan H. Venezia (VA Loma Linda Healthcare System, Loma Linda, CA), and Sebastian Waz (Dept. of Cognit. Sci., Univ. of California, Irvine, CA)

Acoustics research involving human participants typically takes place in specialized laboratory settings. Listening studies, for example, may present controlled sounds using calibrated transducers in sound-attenuating or anechoic chambers. In contrast, remote testing takes place away from the lab, in natural settings or in participants' homes. Remote testing can potentially provide greater access to participants, larger sample sizes, and enhanced ecological validity, at the cost of reduced acoustical control, standardization, calibration, and consistency of participant experiences. Emerging technologies can ameliorate some drawbacks, and potentially support new forms of robust research via remote testing. The ASA Technical Committee on Psychological and Physiological Acoustics (P&P) launched the Task Force on Remote Testing in May 2020, with goals of (1) surveying approaches and platforms available to support remote testing by ASA members, (2) identifying challenges and considerations for prospective investigators, and (3) communicating this information via online resources, papers, and presentations. Longer-term goals include identifying best practices and providing resources for evaluating outcomes of remote testing to facilitate via peer review. This presentation will describe the activities of the P&P Task Force on Remote Testing, online resources identified and/or developed by Task Force members, and additional opportunities for ASA members to contribute.

**4aPPb2. Evaluating the impact of external noise on psychoacoustic testing using the portable automated rapid test (PART) platform.** Dana Cherri (Commun. Sci. and Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, dcherri@mail.usf.edu), Erol J. Ozmeral, and David A. Eddins (Commun. Sci. & Disord., Univ. of South Florida, Tampa, FL)

The Portable Automated Rapid Test (PART) platform consists of a consumer electronics tablet and headphones with software for assessing auditory perception. One goal of PART is to expand our knowledge of the hearing profile of individuals. The standard PART battery includes psychophysical measures that span multiple auditory perceptual domains. To date, most of the psychophysical measures in PART have been validated in quiet laboratory settings. It is unknown whether or how ambient or external noise will impact threshold measures. Subjects completed the PART battery in quiet over headphones (65 dB SPL) and in the presence of external cafeteria noise through loudspeakers (59 dB SPL; consistent with literature on waiting area noise). In separate conditions, we mixed pink noise through the headphones as a means to potentially mitigate the impacts of external noise (50 dB SPL; maximum level that did not impact thresholds). Threshold measures were collected from 10 young listeners with normal hearing. Results indicate that neither the pink noise nor the cafeteria noise affected average thresholds on any of the tasks relative to the quiet condition. These data support the use of PART in relatively uncontrolled listening environments like clinic waiting rooms or at home. [Work supported by NIH R01DC015051.]

**4aPPb3. Anonymous multipart web-based psychoacoustics: Infrastructure, hearing screening, and comparison with lab-based studies.** Brittany A. Mok (Speech, Lang., and Hearing Sci., Purdue Univ., West Lafayette, IN), Vibha Viswanathan, Agudemu Borjigin, Ravinderjit Singh (Weldon School of Biomedical Eng., Purdue Univ., West Lafayette, IN), and Hari Bharadwaj (Speech, Lang., and Hearing Sci., Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, hbharadwaj@purdue.edu)

Web-based experiments offer the potential to collect large datasets from diverse cohorts of listeners, and can help circumnavigate constraints on in-person testing placed by the COVID-19 pandemic. Here, we (1) outline our infrastructure for multipart web-based hearing studies, (2) describe our approach to screening participants for headphone use and "normal-hearing" status, and (3) compare performance trends in the same task paradigms between lab- and web-based studies. Browser-based psychoacoustic tasks were implemented using jsPsych, a free and open-source JavaScript library. Dynamic sequences of psychoacoustic tasks were combined with consent pages, questionnaires, and debriefing pages using Django, a free and open-source Python library for web applications. Subjects were recruited, pre-screened for demographic characteristics, and compensated anonymously via Prolific.co, a web-based human-subject marketplace. Headphone use was checked by supplementing the procedures in Woods *et al.* (Atten Percept Psychophys, 2017) with a binaural hearing task. Guided by a

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meta-analysis of normative data, screening for (near) normal-hearing status (and compliance with instructions) was done by combining scores in a suprathreshold cocktail-party task with survey responses. Individuals meeting all criteria were anonymously re-invited to complete the main psychoacoustic tasks. Preliminary results suggest that performance trends and "main-effects" are comparable to lab-based data.

**4aPPb4. Challenges in the development of a fully autonomous auditory training and testing system.** Douglas S. Brungart (Walter Reed National Military Medical Ctr., 8901 Wisconsin Ave., Bethesda, MD 20889, douglas.s.brungart.civ@mail.mil), Mary Barrett, Shoshannah Russell (Walter Reed National Military Medical Ctr., Bethesda, MD), Véronique Archambault-Léger (Creare, Hanover, NH), Rebecca R. Holtzman, Coral Dirks (Walter Reed National Military Medical Ctr., Bethesda, MD), and Odile Clavier (Creare, Hanover, NH)

The Computerized Auditory Training (CAT) pilot study is a feasibility study utilizing android tablets to remotely administer auditory training and conduct auditory assessments. The goal of the study is to examine user acceptability and the effectiveness of different types of auditory training modules. The CAT protocol administers a predetermined regimen of auditory training sessions interleaved with structured auditory assessments that are designed to track subject progress over time. Participants choose the type of training they want to perform from options that include both interactive game-based training modules and more conventional modules based on standard adaptive psychophysical procedures. Timers are utilized to limit the amount of training that can be performed in a single session. Here, we will discuss some of the challenges in the development of the system and the strategies employed to address them, including (a) the ability to maintain control over the level and frequency response of the acoustic signal presented to the listener; (b) the ability to monitor the background noise level during training and testing; (c) the ability to remotely track participant progress and provide incremental payments for subject participation; and (d) the ability to remotely access and analyze participant data. The views expressed here are those of the author and do not reflect the official policy of the Department of Army, Department of Defense, or U.S. Government.

**4aPPb5. TeamViewer software can facilitate remote data collection for studies initially designed for in-lab testing.** Julie I. Cohen (Hearing and Speech Sci., Univ. of Maryland, 0100 Lefrak Hall, College Park, MD 20742, jcohen6@umd.edu), Sandra Gordon-Salant (Hearing and Speech Sci., Univ. of Maryland, College Park, MD), and Douglas S. Brungart (Audiol. and Speech Pathol., Walter Reed NMMC, Bethesda, MD)

During speech perception experiments, well-controlled and calibrated stimuli are presented to the listener, whose task is often to repeat the speech stimulus heard. Researchers typically score the listener's responses in real-time, and may also record the verbal responses for scoring verification. Due to the COVID-19 pandemic, data collection on an ongoing speech perception study was halted. Our goal was to resume data collection with exactly the same stimuli and procedures while maintaining the safety of the participants and examiner. To that end, TeamViewer (v. 15.7.7) software was used to administer the experiment using remote access to a research laptop. All study materials, including a laptop with the programmed experiment (MATLAB 2011) and high quality headphones were delivered to the participant's home. Stimuli developed for the experiment were recorded remotely by talkers using a microphone and recorder. A custom MATLAB script was run from the experimental laptop to facilitate collection of speech perception data. Both portions of the experiment required the use of the TeamViewer software, to instruct the participant in the experimental protocol and permit oversight of data collection. Specific methods used to conduct stimulus recordings and collect data will be discussed.

**4aPPb6. Feasibility of virtual telehealth cochlear-implant testing.** Joshua G. Bernstein (National Military Audiol. & Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., 4954 N Palmer Rd., Bethesda, MD 20889, joshua.g.bernstein.civ@mail.mil), Elicia M. Pillion, Sandeep A. Phatak, Coral Dirks (National Military Audiol. & Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., Bethesda, MD), and Anthony M. Tolisano (Dept. of Otolaryngol. – Head and Neck Surgery, Walter Reed National Military Medical Ctr., Bethesda, MD)

The COVID-19 pandemic has accelerated remote-testing needs for cochlear-implant (CI) recipients. Progress toward the development and validation of a telehealth system for CI patient care is described. The proposed system contains the most critical elements for a CI programming visit: (1) direct connection via clinical software to evaluate and manipulate electrical-stimulation parameters; and (2) sound-field audiometry and speech-perception testing. The former represents a low technological hurdle as the audiologist remotely controls a laptop preloaded with clinical software that has been mailed to the patient. The latter utilizes a calibrated headphone-based replacement for sound-field testing based on our research experience with simulated free-field listening delivered to the sound processor. This proof-of-concept protocol will include five initial patients in a simulated telehealth environment (audiologist and patient in separate rooms) and 10 patients participating from home. We highlight technological and logistical hurdles overcome, patient and clinician satisfaction, and a comparison of headphone-based to standard-of-care sound-field testing. Implications for remote auditory research with impaired populations are discussed. [Work supported by Defense Health Agency Innovations Group. 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 the U.S. Government.]

**4aPPb7. Automated pure tone audiometry with true wireless stereos earbuds.** Zhenyu Guo (School of Phys. and Optoelectronics, South China Univ. of Technol., No. 23 Bldg., Wushan Rd., Guangzhou 510006, China, zhenyuguo404@qq.com), Xianren Wang (Dept. of Otorhinolaryngology, The First Affiliated Hospital of Sun Yat-sen Univ., Guangzhou, China), Huali Zhou, Yigang Lu, Guangzheng Yu, and Qinglin Meng (School of Phys. and Optoelectronics, South China Univ. of Technol., Guangzhou, China)

Amounts of automated pure tone audiometer applications have been developed for personal terminals in recent years. However, most of them require specifically designed headphones, which are usually expensive and not accessible to most people. The commercially available true wireless stereos earbuds (e.g., Honor Flypods and Apple Airpods) got a prevalence in recent years. In this study, Flypods were used in a specifically developed automated audiometer whose algorithms and calibrations were modified for wireless earphones. The calibration was established by using both a KEMAR manikin and a loudness comparison method. Twenty subjects with mild to moderate hearing loss were recruited in a clinical verification experiment. The average deviation of automated audiometry was 3.1 dB, when referring to the thresholds measured by a conventional manual audiometer with a pair of TDH 39 headphones. Most (63.2%) of the deviations were under 5 dB. The sensitivity and specificity across frequencies from 125 to 8000 Hz were 95.4% and 62.6%, respectively. The results demonstrate that it is practical to utilize this sort of affordable wireless earphones for preliminary hearing level screening.

**4aPPb8. Suitability of self-recordings and video calls: Vowel formants and nasal spectra.** Valerie Freeman (Dept. of Comm Sci & Disord., Oklahoma State Univ., 042 Murray Hall, Stillwater, OK 74078, valerie.freeman@okstate.edu), Paul De Decker (Linguist., Memorial Univ., St. John's, NF, Canada), and Molly Landers (Commun. Sci. & Disord., Oklahoma State Univ., Stillwater, OK)

When the COVID-19 pandemic halted in-person data collection, many linguists adopted new online technologies to replace traditional methods. These included VoIP video conferencing apps like Zoom, which allow live interaction with participants, as well as user-led options in which participants record themselves using personal computers or smartphones and then email or otherwise transfer the sound files to researchers online. This study evaluated the suitability of such recordings for phonetic analysis of vowel

space configurations, mergers, and nasalization by comparing simultaneous recordings from several popular personal devices (Macbook, PC laptop, iPad, iPhone, Android phone) and video call apps (Zoom, Skype, Teams) to those taken from professional equipment (H4n field recorder, Focusrite with table-top microphone). All personal devices and apps conveyed vowel arrangements and nasalization patterns relatively faithfully (especially laptops), but absolute measurements varied, particularly for the female speaker and in the 750-1500 Hz range, which affected the locations (F1xF2) of low and back vowels and reduced nasalization measurements (A1-P0) for the female's pre-nasal vowels. Based on these results we assess the validity of remote recording using these devices and offer recommendations for best practices for collecting high fidelity acoustic phonetic data from a distance.

**4aPPb9. Telepractice: Acoustic characteristics of the patient, parent, and provider speech.** Maria V. Kondaurova (Psychol. & Brain Sci., Univ. of Louisville, 317 Life Sci. Bldg., Louisville, KY 40292, maria.kondaurova@louisville.edu), Cheryl W. Donaldson (The Heuser Hearing Inst. & Lang. Acad., Louisville, KY), Qi Zheng (Biostatistics, Univ. of Louisville, Louisville, KY), Joneen Lowman (Commun. Sci. and Disord., Univ. of Kentucky, Louisville, KY), and Matthew L. Bush (Otolaryngol. – Head and Neck Surgery, Univ. of Kentucky Medical Ctr., Lexington, KY)

How does the use of telepractice during speech therapy affect the acoustic characteristics of the speech-language pathologist (SLP), the child and her parent speech compared to those in conventional in-person intervention? This study examined prosodic and lexical characteristics in an SLP, mothers and their children with cochlear implants (CI) (n = 7, mean age 59 months, range 43–81 months) productions during one 30-min in-person session and one sequential tele-session, order counterbalanced. Mean pitch (Hz), pitch range (Hz), utterance duration (s), speech rate (syll/utterance duration) and mean length of utterance (MLU) were measured in 30 utterances produced by each child, mother and the SLP in the in-person and tele-sessions. Preliminary analysis indicates that the child, her mother and the SLP produced similar mean pitch across both sessions, however, their pitch range was more expanded during the in-person than the tele- session. The SLP and the mothers' utterance durations were longer during the in-person than the tele-session. The child, her mother and the SLP produced faster speech rate and a higher MLU during in-person than the tele- session. Results suggest that acoustic and lexical (MLU) characteristics of the child, the caregiver and the provider speech are affected by telepractice.

THURSDAY MORNING, 10 DECEMBER 2020 — 11:15 A.M. TO 12:00 NOON

## Session 4aPPc

## Psychological and Physiological Acoustics and Speech Communication: Remote Testing for Auditory and Speech Research II (Poster Session)

Authors will be at their posters from 11:15 a.m. to 12 noon.

### *Contributed Papers*

**4aPPc1. Normative psychoacoustic data using portable automated rapid testing (PART) iPad application.** Nirmal Kumar Srinivasan (Dept. of Speech-Lang. Pathol. and Audiol., Towson Univ., 8000 York Rd., Towson, MD 21252, nsrinivasan@towson.edu), Alyssa Pfaffe (Dept. of Speech-Lang. Pathol. and Audiol., Towson Univ., Towson, MD), and Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR)

Psychoacoustic tasks have been used to evaluate auditory processing capabilities throughout many decades. Here, we present data for Spatial Release from Masking (SRM), gap detection thresholds (GDTs), and spectrotemporal modulation (STM) sensitivity on a large group of listeners varying in age and hearing acuity collected using the PART iPad application (https://braingamecenter.ucr.edu/games/p-a-r-t/). Coordinate Response Measure (CRM) sentences were used to quantify SRM. A 4 Hz-2 c/o spectrotemporally modulated broadband noise was used to quantify STM sensitivity. Broadband Gaussian noise was used to measure gap detection thresholds. Initial analyses revealed similar thresholds between traditional methods found in the literature and data collected through the PART application for all psychoacoustic tasks. The ease with which the data is collected using PART application is expected to aid clinicians in rapidly characterizing difficulties perceived by individuals in everyday listening scenarios and in evaluating patient progress with hearing aid adjustments and aural rehabilitation over time.

**4aPPc2. How do listeners use context frequencies in tone-scramble tasks? Evidence from a web-based experiment.** Sebastian Waz (Cognit. Sci., Univ. of California, Irvine, 2201 Social & Behavioral Sci. Gateway, Irvine, CA 92697-5100, swaz@uci.edu) and Charles Chubb (Cognit. Sci., Univ. of California, Irvine, Irvine, CA)

When listeners try to discriminate rapid pure-tone sequences ("tone scrambles") whose frequencies come from a *major* set ($G_5$, $B_5$, $D_6$, and $G_6$) versus a *minor* set ($G_5$, $B\flat_5$, $D_6$, and $G_6$), most ($\approx$70%) perform near chance (Chubb *et al.*, 2013). The rest perform near ceiling. The variation in performance is largely explained by possession of a single processing resource (Dean and Chubb, 2017). Does this resource grant acuity for the target frequencies ($B_5$ and $B\flat_5$) selectively, or does it underlie some context-sensitive ($G_5$, $D_6$, and $G_6$) musical ability? To investigate, we asked listeners to

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perform the discrimination as stimuli were translated randomly from trial to trial in log frequency. Listeners thus needed to use context to succeed. Performance remained well above chance in high-performers. Similar results were found when targets were randomly selected on each trial to be major/minor *thirds* ($B_5$/$B\flat_5$) or *sixths* ($E_6$/$E\flat_6$). We also investigated whether the context tones caused spectral masking of the targets by manipulating the frequency width of the context. We found limited evidence for such masking. The single-resource model explained 66.53% of the variation across all conditions. Since the experiment was administered online, our results may offer recommendations for future web-based psychoacoustic experiments.

**4aPPc3. Basic auditory skills evaluation battery for online testing of cochlear implant users.** Valeriy Shafiro (Commun. Disord. and Sci., Rush Univ. Medical Ctr., 600 South Paulina St., 1015 AAC, Chicago, IL 60612, Valeriy_Shafiro@rush.edu), Megan Hebb, Jasper Oh, Stanley Sheft (Commun. Disord. and Sci., Rush Univ. Medical Ctr., Chicago, IL), Kara J. Vasil, and Aaron C. Moberly (Otolaryngol., The Ohio State Univ., Columbus, OH)

Cochlear implant (CI) performance varies considerably across domains of auditory function, but clinical testing is typically restricted to speech intelligibility. Although other domains, including environmental sound and music perception or speech comprehension, are considered important to quality-of-life and personal safety, their adaptation into clinical use has been limited. The basic auditory skills evaluation (BASE) battery was developed to provide online access to such comprehensive assessment using the Internet as a test delivery platform. It includes 17 3–5 min tests to evaluate (1) basic spectro-temporal processing, (2) processing of environmental sounds and music, and (3) speech perception and comprehension in quiet and background noise. Two groups of 18 and 25 postlingual adult CI users were tested either at home following online instructions or in the lab by an audiologist. Results revealed a range of performance across tests from near ceiling to close to chance. No significant differences in group performance on any of the tests were observed, suggesting that BASE battery can assess a diverse array of abilities online, with varying difficulty levels. These findings provide preliminary support for the use of the test battery for an Internet-based comprehensive assessment of ecologically relevant aspects of auditory perception in adult CI users.

**4aPPc4. Evaluating tests of binaural hearing for hearing impaired listeners using a portable device.** Anna C. Diedesch (Commun. Sci. & Disord., Western Washington Univ., 516 High St., MS 9171, Bellingham, WA 98225, anna.diedesch@wwu.edu), Sheng Jie Adelaide Bock (Commun. Sci. & Disord., Western Washington Univ., Bellingham, WA), and Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR)

Individuals with sensorineural hearing loss may have similar monaural thresholds (i.e., pure tone audiogram) but perform quite differently on tests of speech-in-noise. Additional testing using complex, binaural signals may be able to differentiate individuals with similar pure-tone hearing. Here, individuals with mild-to-moderate hearing losses were evaluated using tests of spectral, temporal, and spectrotemporal modulation, diotic and dichotic frequency modulation, gap detection, 2 kHz tone-in-noise, and spatial release from masking. Binaural psychoacoustic measurements were evaluated on an iPad using the Portable Automated Rapid Test (PART) app. Normal hearing listeners were evaluated as controls. Normal hearing results were similar to normative data previously collected by Lelo de Larrea-Mancera and colleagues (2020) and hearing-impaired results were elevated for most binaural measurements evaluated compared to normal hearing controls. Binaural psychoacoustic data were successfully measured for individuals with mild-to-moderate amounts of hearing loss using a portable system. Results show that evaluating complex stimuli on clinical populations using a portable device is feasible. Additionally, if accomplished on a larger scale the current test battery may likely add useful information on how hearing-impaired listeners process complex binaural stimuli and could compliment current clinical diagnostic test battery and, potentially, guide hearing aid fittings.

**4aPPc5. Examining spectral and intensity resolution in web-based vocoder simulations of electric hearing.** Jordan A. Beim (Psych., Univ. of Minnesota, N218 Elliott Hall, 75 East River Rd., Minneapolis, MN 55455, beimx004@umn.edu), Heather Kreft, and Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN)

The poor spectral resolution in current cochlear implants contributes to the difficulty implant users face when trying to listen in noisy backgrounds. Recent work examining measures of spectral resolution and speech perception in normal-hearing and cochlear-implant listeners found that the amount of simulated spectral resolution producing similar speech-in-noise performance between groups resulted in poorer performance for the normal-hearing group on spectral-ripple detection and discrimination tasks. The results suggest that simulating poorer spectral resolution alone is not sufficient to explain implant user performance. Spectral-ripple detection performance likely reflects both spectral- and intensity-resolution, but few vocoder simulations utilize implant-like intensity resolution. In this study, a vocoder processing strategy simulating both electric field spread and dynamic range limitations was used to evaluate performance of normal-hearing and cochlear-implant listeners on speech-in-noise and spectral-resolution tasks. Data were collected using a novel online remote testing platform developed implemented with the MATLAB Web App Server. A comparison of results between groups highlights the implications of including psychophysically measured estimates of cochlear-implant performance in acoustic simulations of electric hearing. [Work supported by NIH grant R01DC012262.]

**4aPPc6. Enhanced loudness perception in ears with reduced peripheral function: A remote psychoacoustic investigation.** Kelly N. Jahn (Eaton-Peabody Labs., Massachusetts Eye and Ear, 243 Charles St., Boston, MA 02114, kelly_jahn@meei.harvard.edu), Kenneth E. Hancock, Stéphane F. Maison, and Daniel B. Polley (Eaton-Peabody Labs., Massachusetts Eye and Ear, Boston, MA)

Central gain models of hyperacusis propose that a loss of afferent input drives over-amplification of sound-evoked activity in the central auditory system, leading to enhanced loudness perception. We conducted a remote study aimed at quantifying variation in loudness perception as a function of peripheral auditory status [i.e., summating potential (SP)/action potential (AP) ratio]. To control for the contributions of age and other individual differences, twelve adults with normal hearing and strongly asymmetric SP/AP ratios between ears were recruited from a prior study. Participants completed a battery of self-directed psychoacoustic tests at home using calibrated headphones and tablet computers running custom software. Loudness perception was assessed at a variety of frequencies in each ear using standard clinical assessments [e.g., loudness discomfort levels (LDLs), dynamic range] and more fine-grained assessments of loudness growth as a function of stimulus intensity. While traditional measures of LDL and dynamic range varied widely both within and between subjects, ears with evidence of possible cochlear synaptopathy (high SP/AP ratios) demonstrated elevated loudness perception for intensities in-between threshold and LDL. Detailed assessments of loudness growth across sensation levels may be more sensitive to variation in underlying peripheral neural status than traditional, single-point clinical measures of loudness discomfort.

**4aPPc7. Telepractice in pediatric speech therapy: Characteristics of patient, parent, and provider vocal interaction.** Maria V. Kondaurova (Psychol. & Brain Sci., Univ. of Louisville, 317 Life Sci. Bldg., Louisville, KY 40292, maria.kondaurova@louisville.edu), Cheryl W. Donaldson (The Heuser Hearing Inst. & Lang. Acad., Louisville, KY), Abigail L. Betts (Otolaryngol., Head/Neck Surgery & Communicative Disord., Univ. of Louisville, Louisville, KY), Qi Zheng (Biostatistics, Univ. of Louisville, Louisville, KY), Joneen Lowman (Commun. Sci. and Disord., Univ. of Kentucky, Louisville, KY), and Matthew L. Bush (Otolaryngol. – Head and Neck Surgery, Univ. of Kentucky Medical Ctr., Lexington, KY)

How does the use of modern technology, such as telepractice, affect the characteristics of vocal interaction between the speech-language pathologist (SLP), the child, and the caregiver? This study examined the prevalence and temporal structure of vocal turns between the SLP, mothers and their children with cochlear implants (CI) ($n = 7$, mean age 59 months, range 43-81

months) during one 30- minutes in-person and one sequential tele session, order counterbalanced. The frequency of vocalizations, vocal turns, simultaneous speech and between-speaker pause duration in tele versus in-person sessions were examined. Results indicate that the SLP vocalized more during in-person than the tele session; mothers demonstrated an opposite result. The child vocalization frequency was not depended on session type. There were more turns between the SLP and the child during in-person than tele session; opposite results were found for mother-child turns. The number of turns between the SLP and the mother and the occurrence of simultaneous speech were not affected by session type. Pauses were longer between the SLP and the child and between the child and her mother in tele- than in-person session. Results suggest that vocal interactions between the patients, parents and providers are impacted by the intervention delivery modality.

**4aPPc8. Speech data collection at a distance: Comparing the reliability of acoustic cues across homemade recordings.** Cong Zhang (Univ. of Kent, Cornwallis North West, Canterbury CT2 7NF, United Kingdom, congzhang.oxford@gmail.com), Kathleen Jepson, Georg Lohfink (Univ. of Kent, Canterbury, United Kingdom), and Amalia Arvaniti (Radboud Univ., Nijmegen, Netherlands)

Speech production data collection has been significantly impacted by COVID-19 restrictions. Sound-treated recording spaces and high-quality recording devices are inaccessible, and face-to-face interactions are limited. We investigated alternative recording methods that produce data suitable for phonetic analysis, and are accessible to people in their homes. We examined simultaneous recordings of pure tones at seven frequencies (50 Hz, every 100 Hz between 100 Hz and 600 Hz), and three repetitions of the primary cardinal vowels elicited from five trained speakers. Recordings were made using the ZOOM meeting application and non-lossy format smartphone applications (Awesome Voice Recorder, Recorder), comparing these with Zoom H6N reference recordings. F0, F1-5, and duration based on manual segmentation were measured. F0 is highly correlated between the three devices for vowels and tones. Lower formants are also significantly correlated though not as robustly. The upper formants showed more variation as reported in the literature. Both phone and ZOOM performed better for vowels than tones. Phone segmentation generated reliable duration values differing from H6N segmentation by $\sim$18 ms. However, irregular waveforms and filtering algorithm artefacts caused considerable differences for ZOOM ($\sim$119 ms). Our preliminary study suggests phone recordings are a viable option for some phonetic studies (e.g., prosody). Future analysis of natural speech data will prove insightful.

**4aPPc9. Identification of unreliable subject data from remote testing.** Daniel E. Shub (Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., Bldg. 19, Rm. 5400, 4954 North Palmer Rd., Bethesda, MD 20889, daniel.e.shub.civ@mail.mil), Trevor T. Perry, Matthew J. Makashay (Army Hearing Program, U.S. Army Public Health Ctr., Aberdeen, MD), Hector Galloza (2GaRi Solutions, Inc., Aguada,), and Douglas S. Brungart (Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., Bethesda, MD)

The less controlled nature of conducting behavioral testing outside of the typical sound booth and not under the direct observation of the experimenter introduces additional factors that may lead to unreliable data. These factors arise from the environment (e.g., room acoustics and background noise), the participant (e.g., attention, engagement, and understanding of the task), and the testing apparatus (e.g., calibration and network latency). Remote-testing protocols should include features to allow for the detection of unreliable data. To aid in the detection of unreliable data in future remote-testing protocols, we retrospectively examined a large dataset ($N > 1000$) of speech intelligibility and tone-in-noise detection that was collected over headphones with a tablet-based system in a waiting room environment. The dataset provides the ability to determine the extent to which unreliable data can be detected based on the presence of high levels of background noise during the testing, abnormally short response times, high lapse rates, flat psychometric functions, and patterns in the response sequence. [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.]

# Session 4aSAa

## Structural Acoustics and Vibration, Signal Processing in Acoustics, Noise, and Engineering Acoustics: Active or Tunable Structural Acoustics I

Christina J. Naify, Cochair
*Naval Research Lab, 4555 Overlook Ave. SW, Washington, D.C. 20375*

Benjamin S. Beck, Cochair
*Marine and Physical Acoustics, Penn State Applied Research Lab, PO Box 30, MS 3200D, State College, PA 16804*

**Chair's Introduction—9:30**

### *Invited Paper*

**9:35**

**4aSAa1. Smart panel with piezoelectric patches connected to adaptive multi-resonant shunts for broadband vibration control.** Paolo Gardonio (DPIA, Univ. of Udine, Via Delle Scienze, 206, Udine 33100, Italy, paolo.gardonio@uniud.it), Gabriel Konda Rodrigues, Loris Dal Bo, and Emanuele Turco (DPIA, Univ. of Udine, Udine, Italy)

This paper presents a simulation study on the broadband control of flexural vibration of thin panels equipped with piezoelectric patches connected to adaptive multi-resonant shunts. The study considers the broadband flexural response of a thin rectangular aluminum panel exposed to a spatial and time stochastic excitation. The panel is equipped with thin square piezoelectric patches connect to shunts encompassing multiple parallel branches formed by RLC components connected in series. The capacitance C of each branch is kept fixed, whereas the inductance L and resistance R are adapted so as each branch maximizes the vibration power absorption from the resonant response of a specific flexural mode of the panel. The study proposes an on-line tuning approach, where the N-branches of each shunt are adapted sequentially to maximize the vibration absorption from the resonant responses of N neighbor flexural modes. The vibration power absorption of each branch is estimated from the electric power absorbed by the shunt. In this way a local tuning system can be implemented, which adapts on-line the shunts to control the flexural response produced by a group of flexural modes of the panel resonating in a given frequency band.

### *Contributed Papers*

**9:55**

**4aSAa2. Nonlinear acoustic-structure interaction in high intensity focused ultrasound power transfer systems.** Aarushi Bhargava (Dept. of Radiology, Univ. of Chicago, 208 Broce Dr., Apt. 1, Blacksburg, VA 24060, aarushi@vt.edu), Vamsi C. Meesala (Biomedical Eng. and Mech., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA), and Shima Shahab (Mech. Eng., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA)

Ultrasound power transfer (UPT) has emerged as one of the most promising technique to wirelessly transfer power. The mechanism of UPT involves vibration-induced generation of acoustic waves from a piezoelectric transmitter which propagate in a medium to interact with a piezoelectric receiver, and generate vibrations-induced electrical response in the receiver. In this work, a novel technique of using a high intensity focused ultrasound (HIFU) is proposed to enhance the power transfer efficiency as compared to current UPT systems. HIFU concentrates the entire transmitted energy at a focal spot where the piezoelectric receiver is placed, thus dramatically reducing the input energy required, as compared to unfocused UPT systems. The physics of the HIFU-UPT system consists of a nonlinear acoustic wave propagating in a medium to induce a nonlinear structural response in the receiver. Implementing the HIFU-UPT phenomena in an experimentally validated finite-element based model shows that the nonlinear acoustic field can induce disproportionately high responses in a receiver if the harmonics in the acoustic wave coincide with the structural resonant frequencies of the receiver. This research aims to provide a comprehensive framework for designing UPT systems operating in nonlinear acoustic fields with a finite size receiver.

**10:15**

**4aSAa3. Diffusive subsurface propagation of Rayleigh mode induced by embedded ultrasonic waveguide.** Yishi Lee (Elec. and Comput. Eng., Univ. of Denver, 2155 E. Wesley Ave., Denver, CO 80210, leeyishi@gmail.com)

Surface degradation in the wood surface has generated unwanted variation in the ultrasound and reduced the efficacy of ultrasonic based non-destructive evaluation (NDE). This study proposes a novel alternative by inserting a waveguide into the wooden medium to eliminate the impact of surface variation on the characteristics of the ultrasonic propagation. This technique creates a small circular insertion of 2 cm depth with 5 mm in diameter, which has been accepted by many utility inspection firms for structural health monitoring (SHM) of wooden utility grids. This work develops a closed-form analytical solution of the elastodynamic problem using the proposed embedded waveguide technique. The resulted solution based on the Navier's formulation discovered two interfering wave modes embedded in the displacement field. These two modes termed *fast* and *slow* induce diffusive displacement propagation as a function of the Poisson's ratio of the medium. The analytical result also shows a greater elastic energy penetration as a function of the insertion depth. Resulted wave characteristics are different from the traditional contact-based Rayleigh excitation. A developed numerical model using the semi-explicit differential-algebraic equation (DAE) technique with both steady-state and transient load conditions is demonstrated. The simulated displacement fields are analyzed and compared with the analytical result.

## Session 4aSAb

# Structural Acoustics and Vibration, Signal Processing in Acoustics, Noise, and Engineering Acoustics: Active or Tunable Structural Acoustics II

Christina J. Naify, Cochair
*Naval Research Lab, 4555 Overlook Ave. SW, Washington, D.C. 20375*

Benjamin S. Beck, Cochair
*Marine and Physical Acoustics, Penn State Applied Research Lab, PO Box 30, MS 3200D, State College, PA 16804*

**Chair's Introduction—11:15**

### *Contributed Papers*

**11:20**

**4aSAb1. Experimental determination of wavenumber dispersion for active adaptable acoustic metamaterials.** Aaron Stearns (Mech. Eng., Penn State Univ., P.O. Box 30 Burrowes St., State College, PA 16804-0030, ajs6037@psu.edu) and Benjamin S. Beck (Marine and Physical Acoust., Penn State Appl. Res. Lab, State College, PA)

Acoustic metamaterials are composite materials exhibiting effective properties and acoustic behavior not found in traditional materials. Through periodic subwavelength resonant inclusions, acoustic metamaterials enable steering, cloaking, lensing, and frequency band control of acoustic waves. A common drawback of acoustic metamaterials is that the properties are limited to narrow frequency bands. Investigation of practical active and adaptable acoustic metamaterials is valuable in achieving wider operation frequency bands. Numerical predictions of wave propagation behavior in acoustic metamaterials are commonly presented in the form of elastic band structure diagrams. In previous work, the complex wavenumber dispersion properties of the metamaterial medium were proposed as optimization objectives for obtaining optimal adaptable metamaterial unit cell configurations for vibration reduction. To verify numerical wavenumber predictions, the current work presents an experimental method to obtain the wavenumber dispersion. First, the metamaterial beam is excited with a broadband pulse. Time domain responses are recorded at many locations on the structure. The resulting time series data is processed with a two dimensional Fourier transform. The result is a wavenumber versus frequency plot. The procedure is useful for plate and beam type metamaterial structures wherein local resonances and active inclusions cause wave attenuation if the above experimental procedures can be feasibly carried out.

**11:40**

**4aSAb2. Tunable vibration control in a microelectromechanical resonator array by means of parametric coupling between mechanical and electrical modes.** Sushruta Surappa (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, Atlanta, GA 30313, sushsurappa@gatech.edu) and F. Levent Degertekin (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Vibration control in flexible structures at both the micro and macro scale has been a subject of considerable research interest over the last few decades. Various active, semi-active and passive vibration control schemes have been proposed for applications ranging from attenuating unwanted vibrations in structural components to controlling the ring down time in high-Q mechanical oscillators. Here, we present a simple, yet highly effective method to passively attenuate large amplitude resonant vibrations in an array of membrane-based microelectromechanical (MEM) resonators. By capacitively coupling the MEM array to a resonant electrical circuit tuned to half the mechanical resonance frequency, the vibrational energy in the array can be converted to electrical energy via parametric mode coupling between the mechanical and electrical resonator. Simulation and experimental studies indicate that both the rate of vibration attenuation and the degree of suppression can be effectively controlled by tuning electrical parameters such as the resistance and the initial voltage in the circuit. The extension of the control scheme to multi-mode vibrations by making use of multiple shunted electrical resonators will also be briefly discussed.

**12:00**

**4aSAb3. Switchable acoustic metamaterial design through instability-driven microstructure transformations in soft composites.** Stephan Rudykh (Mech. Eng., Univ. of Wisconsin-Madison, 77 Mass. Ave., 1-025, Cambridge, MA 02139, rudykh@mit.edu)

Soft materials can sustain large deformations, and, thus, can be used as a platform for tunable and switchable systems capable of wave manipulations. The fascinating elastic instability phenomenon in soft materials can be used to design new (meta-) materials with switchable microstructures, properties, and functions. Examples include the emergence of tunable band gaps at low-frequency ranges, the appearance of negative group velocity, and extreme wave slowing-down. Here, we investigate the elastic instability phenomenon in soft heterogeneous materials, and the applications of the phenomenon to design of soft acoustic metamaterials. The deformable composites typically combine soft matrix and stiffer phases (such as fibers or inclusions). We will start by considering the effect of experimentally observed wrinkling on elastic waves band-gaps in soft laminates. Next, we will discuss the emergence of the negative group velocity in 3-D fiber composite brought to the "marginally stable" state. Then, we will turn to the so-called auxetic or negative Poisson's ratio materials comprising of soft-matrix-void-stiff-inclusion systems, and illustrate the mechanisms leading to the emergence of low-frequency band gaps. Finally, a new type of instabilities giving rise to anti-symmetric domain formations will be illustrated on the examples of 3-D printed soft composites.

## Session 4aSCa

# Speech Communication, Architectural Acoustics, Psychological and Physiological Acoustics, and Noise: Listening in Challenging Circumstances V (Poster Session)

Authors will be at their posters from 9:30 a.m. to 10:15 a.m.

### *Contributed Papers*

**4aSCa1. More “rhythmic” speech is more intelligible in noise: Evidence from Lombard-inspired speech modifications.** Hans Rutger Bosker (Max Planck Inst. for Psycholinguistics, P.O. Box 310, Nijmegen 6500 AH, The Netherlands, HansRutger.Bosker@mpi.nl) and Martin Cooke (Lang. and Speech Lab., Universidad del País Vasco, Vitoria-Gasteiz, Spain)

When listening in challenging circumstances, such as in loud background noise, speakers may adjust their voice, known as Lombard speech. These acoustic adjustments facilitate speech comprehension in noise relative to plain speech (i.e., speech produced in quiet). However, exactly which characteristics of Lombard speech drive this intelligibility benefit in noise remains unclear. This study assessed the contribution of enhanced amplitude modulations to the Lombard speech intelligibility benefit. We demonstrate that (1) across a range of cross-language speech corpora, talkers show greater power in the modulation spectrum for speech produced in noise as compared to RMS-matched speech-in-quiet; (2) more enhanced amplitude modulations (i.e., more ‘rhythmic’ speech) correlate positively with intelligibility in a speech-in-noise perception experiment; (3) transplanting the amplitude modulations from Lombard speech onto plain speech leads to an increase in keywords correct scores, suggesting that enhanced amplitude modulations in Lombard speech contribute towards intelligibility in noise. Results are discussed in light of recent neurobiological models of speech perception with reference to neural oscillators phase-locking to the amplitude modulations in speech, guiding comprehension.

**4aSCa2. Speech perception in deaf and hard of hearing Arabic-speaking children.** Judith Rosenhouse (Lingusitics, SWANTECH, Ltd., 9 Kidron St., Haifa 3446310, Israel, Judith@swantech.co.il)

The rate of Deaf and Hard of Hearing (DHH) people is relatively large in native Arabic speakers in Israel. However, Arabic assessment tools are scarce. The present study reports the procedure of developing two speech perception assessment tools, which are used for hearing loss diagnosis, hearing aids fitting, cochlear implant mapping, and evaluation of progress following rehabilitation intervention. Our closed set test “Arabic Picture Speech Pattern Contrast” (ArPiSPAC), and word perception open-set test, “Arabic AB” (ArAB), were administered to 34 children with moderate to profound sensorineural hearing loss, hearing aids users, aged between 4;5—8;11. In addition, the ArPiSPaC was administered to 38 hearing children, aged between 2;6-5;5 in order to obtain a developmental hierarchy. For DHH participants, the findings showed that vowel contrasts were better perceived than consonants, and articulation manner was better perceived than articulation place. The hearing participants’ phonological perception hierarchy differed in some interesting features from that of the DHH children. The mean percent of phoneme perception by DHH was $M = 60.53\%$, ($SD = 23.87$), and the mean percent of word perception was $M = 18.5\%$, ($SD = 15.68$). Voicing and pharyngealization contrasts revealed the largest Pearson correlation for phoneme and word perception.

**4aSCa3. The effects of word status and vocoding on voice cue perception.** Thomas Koelewijn (Otorhinolaryngology/Head and Neck Surgery, Univ. Medical Ctr. Groningen, Hanzeplein 1, Groningen 9713 GZ, The Netherlands, t.koelewijn@rug.nl), Etienne Gaudrain (Lyon Neurosci. Res. Ctr., CNRS, Lyon, France), Terrin N. Tamati (Otolaryngol., The Ohio State Univ., Columbus, OH), and Deniz Başkent (Otorhinolaryngology/Head and Neck Surgery, Univ. Medical Ctr. Groningen, Groningen, Netherlands)

Talkers’ voices play an important role in speech perception, a process affected in CI-users. While previous research has suggested top-down influence of linguistic content on talker perception, how this affects perception of individual voice cues when vocoded remain unclear. Fifteen young, normally hearing participants performed a three alternative forced choice adaptive task. They listened to words presented normal (forward) or reversed (word status), while for one of the three words the fundamental frequency (F0) and/or vocal-tract length (VTL) was manipulated. In addition, either the same or three different words/reversed-words (token homogeneity) were presented in sequence, without or with (low or high simulated stimulation-spread) vocoding. Results show that JNDs for F0 and VTL were significantly smaller (better) for words compared to reversed-words, for fixed than for variable conditions, and for without than with vocoding. A four-way ANOVA and *post hoc* analysis revealed the word status was mostly affecting VTL JNDs in the variable word conditions. This suggests the phonotactic properties of vowels impact VTL perception. In all, linguistic (phonological) processing seems to have a top-down effect on voice cue perception especially when linguistic variability is high. These cognitive compensation mechanisms might also benefit CI-users, especially when it comes to VTL discrimination.

**4aSCa4. Differences in speech-on-speech processing between musicians and non-musicians: The role of durational cues.** Elif C. Kaplan (Univ. of Groningen/UMCG, Rm. P1.204, Hanzeplein 1, Groningen 9713 GZ, The Netherlands, e.c.kaplan@rug.nl), Deniz Başkent, and Anita E. Wagner (Univ. of Groningen/UMCG, Groningen, The Netherlands)

In the current study, we investigate the role of prosodic cues in speech-on-speech perception in musicians and non-musicians. Earlier studies have shown that musically experienced listeners may have an advantage in speech-on-speech perception performance in behavioral tasks (Baskent & Gaudrain, 2016; Swaminathan *et al*., 2015). Previously, we have also shown in an eye-tracking study, with the visual world paradigm, that musical experience has an effect on the timing of resolution of lexical competition when processing speech in two-talker speech masker (Kaplan *et al*., 2018). In particular, musicians were faster in lexical decision-making when the two-talker speech masker was added to target speech. However, the source of the difference observed between the two groups remained unclear. In the current study, again by employing a visual world paradigm, we aim to clarify whether musicians make use of durational cues that contribute to prosodic boundaries in Dutch, in resolving lexical competition when processing non-masked versus two-talker masked speech. If musical training preserves listeners’ sensitivity to the acoustic correlates of prosodic boundaries when processing masked speech, we expect to observe more lexical competition and delayed lexical resolution in musicians.

**4aSCa5. The benefit from voice gender cue differences for the perception of speech in competing speech in school-age children.** Leanne Nagels (Ctr. for Lang. and Cognition Groningen, Univ. of Groningen, Harmonie Bldg., Oude Kijk in Het Jatstraat 26, Groningen 9712 EK, The Netherlands, leanne.nagels@rug.nl), Etienne Gaudrain (CNRS UMR5292, Lyon Neurosci. Res. Ctr., Inserm U1028, Université de Lyon, Lyon, France), Debi Vickers (Cambridge Hearing Group, Sound Lab, Clinical Neurosciences Dept., Univ. of Cambridge, Cambridge, United Kingdom), Petra Hendriks (Ctr. for Lang. and Cognition Groningen, Univ. of Groningen, Groningen, The Netherlands), and Deniz Başkent (Dept. of Otorhinolaryngology/Head and Neck Surgery, Univ. Medical Ctr. Groningen, Univ. of Groningen, Groningen, The Netherlands)

Differences in speakers' voice gender improve the perception of speech in competing speech for adult listeners, as do differences in only speakers' mean fundamental frequency (F0) or vocal-tract length (VTL), the primary determinants of speakers' perceived voice gender. Children's sensitivity to voice cue differences continues to develop throughout childhood, which may prevent them from benefiting from differences in speakers' voice gender cues during the perception of speech in competing speech. Hence, we investigated how children's benefit from differences in speakers' F0 and VTL cues for perceiving speech in competing speech develops during school-age years (4–12 years). Fifty-five children and fifteen adults participated in a CRM-task with a single-talker speech masker using four masker voice conditions (no F0 or VTL change compared to the target voice, a change in only F0, a change in only VTL, or a change in both F0 and VTL) and three target-to-masker ratios (TMRs). Children's performance improved gradually with age and TMR, but more importantly, children benefited from F0 or VTL differences at all tested ages, even the youngest children. Our results suggest that children can benefit from voice gender cue differences at all tested ages despite their reduced sensitivity to voice cue differences.

**4aSCa6. Release in linguistic masking changes over time: differences between native and non-native listeners.** Alex Mepham (Dept. of Psych., Univ. of York, York YO10 5DD, United Kingdom, am2050@york.ac.uk), Yifei Bi, and Sven Mattys (Dept. of Psych., Univ. of York, York, United Kingdom)

Research into speech-on-speech masking has identified what is known as release from linguistic masking, i.e., better speech transcription performance against a masker in an unknown than known language. To test whether native (English) and non-native (Mandarin) speakers of English could learn to control the interference of a known language masker, we measured their ability to transcribe English sentences against the background of English or Mandarin competing talkers over the course of 50 trials. Both groups improved over time. Native listeners exhibited release from linguistic masking, with less masking in the Mandarin than English masker condition. The size of this effect increased over time. In contrast, non-native listeners showed no difference between the two language maskers, with this pattern unaffected by time. Masker-to-target intrusion errors decreased over time for native listeners, whereas they were virtually absent for the non-native listeners. The results show that (1) Linguistic masking is worst when the masker language is known to the listener, whether that language is native or non-native, (2) Whether the masker is the same language as the target language is comparatively less important for non-native listeners, and (3) Native listeners are better at learning to suppress masker interference over time than non-native listeners.

**4aSCa7. The effect of hearing aid signal processing on speech intelligibility in a realistic virtual sound environment.** Naim Mansour (Dept. of Health Technol., Tech. Univ. of Denmark, Ørsteds Plads Bldg. 352, Kongens Lyngby 2800, Denmark, naiman@dtu.dk), Marton Marschall, Tobias May (Dept. of Health Technol., Tech. Univ. of Denmark, Kongens Lyngby, Denmark), Adam Westermann (WSAudiology, Lynge, Denmark), and Torsten Dau (Dept. of Health Technol., Tech. Univ. of Denmark, Kgs. Lyngby, Denmark)

To develop hearing aids (HA) that benefit users in real-world acoustic conditions, it may be helpful to evaluate HA algorithms psychoacoustically, in controlled yet realistic laboratory settings. However, typical outcome measures like speech intelligibility (SI) are commonly obtained using rather artificial acoustic stimuli. This study investigated the effect of HA signal processing on SI in a more ecologically valid way, using a spatialized version of the Danish Hearing In Noise Test inside a loudspeaker-based virtual sound environment (VSE). The VSE employed spatial recordings of two real-world sound scenarios and quadraphonic speech-shaped noise (SSN) as distinct masker stimuli with spatialized target speech. Unaided performance was compared with results obtained in two master HA configurations: a basic setup consisting of dynamic range compression only and a setup including beamforming as well. Overall, speech reception thresholds (SRTs) were significantly higher in the conditions with real-world noise maskers compared to the artificial SSN masker. HA performance depended strongly on input signal-to-noise ratio (SNR) and was most effective at negative SNRs. The HA setup including beamforming only revealed benefits for the SSN masker. This study underscores the importance of evaluating SI and HAs at realistic SNRs, allowing for better prediction of real-life HA benefit.

**4aSCa8. Processing of prosodically marked focus in a cochlear implant simulation by non-native listeners: Preliminary results.** Marita K. Everhardt (Univ. of Groningen/Univ. Medical Ctr. Groningen, Oude Kijk in 't Jatstraat 26, Groningen 9712 EK, The Netherlands, m.k.everhardt@rug.nl), Anastasios Sarampalis, Matt Coler, Deniz Başkent, and Wander Lowie (Univ. of Groningen/Univ. Medical Ctr. Groningen, Groningen, The Netherlands)

Speakers can use prosodic cues to direct listeners to a specific part of an utterance. The prosodically emphasised part has linguistic focus, determined by the semantic and pragmatic context (e.g., Cole, 2015). For cochlear implant (CI) users, processing prosodically marked focus can be challenging given the degradation in fine spectrotemporal detail of the signal transmitted through the device (e.g., Başkent *et al*., 2016). An additional challenge can be expected for CI users listening to a non-native language. In this ongoing study, we investigate how native Dutch learners of English process prosodically marked focus in English sentences degraded by a CI simulation compared to how they process it in non-CI-simulated stimuli. These results are compared to those of native English listeners. Listeners are presented with English sentences differing in prosodically marked sentential focus and are instructed to indicate which of four possible context questions prompted the response stimulus. We expect that listeners are less accurate and less efficient for the CI-simulated stimuli compared to the non-CI-simulated stimuli and that non-native listeners are less efficient than native listeners, underlining the challenges of prosodically marked sentential focus processing in a non-native language with CI hearing.

**4aSCa9. A real-time wavelet-based algorithm for improving speech intelligibility.** Yijia Chen (Hong Kong Univ. of Sci. and Technol., Rm. 2448, Academic Bldg., Clear Water Bay, Hong Kong, China, ychenfo@connect.ust.hk), Yuxuan Wan, Keegan Y. Sim, Jiakun Zheng (Hong Kong Univ. of Sci. and Technol., Hong Kong, China), Eugene Chau (Lenox Hill Hospital, New York, NY), and Kevin Chau (Hong Kong Univ. of Sci. and Technol., Hong Kong, China)

Presently reported is a wavelet-based algorithm to improve speech intelligibility. The speech signal is split into frequency sub-bands via a multi-level discrete wavelet transform. Various gains (or attenuations) are applied to the sub-band signals before the signals are recombined to form a modified version of the speech signal. Dynamic range compression then follows to control the peak amplitude. The sub-band gains are adjusted while keeping the overall signal energy unchanged, and the speech intelligibility under simulated hearing loss conditions and various background interferences such as babble, music and machine noises is evaluated objectively and quantitatively using Google Speech-to-Text transcription. For both English and Chinese speech, intelligibility is enhanced, and the transcription accuracy can increase by over 60 percentage points by reallocating the audio energy from low to mid and high frequency sub-bands, effectively increasing the consonant-to-vowel intensity ratio. This is reasonable since the consonants are relatively weak and of short duration, and are therefore the most likely to become indistinguishable in the presence of background disturbances or high frequency hearing impairments. The proposed algorithm is implementable in real-time and simpler than others. Potential applications include speech transmission, hearing aids, machine listening, and a better understanding of speech intelligibility.

**4aSCa10. Perception of atomic speech.** Qinglin Meng (Acoust. Lab., School of Phys. and Optoelectronics, South China Univ. of Technol., Bldg. 18, No. 381, Wushan Rd., Tianhe Qu, Guangzhou 510641, China, mengqinglin08@gmail.com)

Natural speech signals are continuous and have great redundancy in time, which is learned by human beings and guarantees speech comprehension even in challenging circumstances. This work develops a sound synthesis method to study some time-related characteristics of the human auditory system. The temporal information within the individual frequency band of a speech is down-sampled using Gaussian-shaped pulses and then recombined into a new sound which may have some remaining intelligibility or feasible phonetic features. The "atomic" sound is coined to the sound with an extremely spectral-temporally sparse spectrogram generated using the method. A battery of speech perception tests was administered in normal-hearing listeners. Results show that (1) atomic sounds from clear speech can be understood as speech, although the listeners often reported a feeling of water sound textures; (2) the temporal and spectral resolution could be traded off in the atomic speech comprehension; and (3) only one maximum envelope value preserved among a 32-channel filter-bank at a rate of 400 Hz was surprisingly adequate for speech understanding, which indicates that the brain can organize the only spectral peak within each short duration no longer than 2.5-ms into sentence understanding without either explicit or implicit encoding of the first three formants.

**4aSCa11. Speech intelligibility improvement of cochlear implant using release of masking.** Dhany Arifianto (Eng. Phys., Inst. Teknologi Sepuluh Nopember, Sukolilo Campus, Surabaya 60111, Indonesia, dhany@ep.its.ac.id)

This research emphasis on an increase in speech intelligibility due to the influence release of masking phenomena as well as testing a variety of enhancement techniques to further improve speech intelligibility. Experiments carried out by listening to the sentence stimuli respondents after those respondents are required to rewrite the stimuli. After that assessment percent correct word for each experiment. Based on research that has been done then some conclusions can be drawn that the influence release of masking has demonstrated an increase in speech intelligibility from the first trial to the second trial of $\pm 7.9\%$. Of the eight signal enhancement techniques are used, Spectral Substraction still showed the best result that is 2.57499 dB in measuring the results of the second experiment a mixture of stimuli 20 and 12.

THURSDAY MORNING, 10 DECEMBER 2020 — 10:15 A.M. TO 11:00 A.M.

# Session 4aSCb

## Speech Communication: Prosody I (Poster Session)

Authors will be at their posters from 10:15 a.m. to 11:00 a.m.

### *Contributed Papers*

**4aSCb1. Cayuga word melodies—A corpus phonetic study.** Richard J. Hatcher (Linguist, Univ. at Buffalo, 378, Niagara Falls Blvd, Buffalo, NY 14223, rjhatche@buffalo.edu)

This work investigates the intonation of Cayuga, an endangered language of North America, now spoken by some 50 speakers in upstate New York and southern Ontario. Word prominence in Cayuga has been described as sensitive to both word shape and utterance position (Foster, 1982; Michelson, 1988). Non-utterance-final words in Cayuga are known to exhibit f0 prominence on the word-final syllable, although prominence occurs on earlier syllables in utterance-final contexts. This work constitutes an initial corpus phonetic investigation into the interaction of word-shape and utterance position on f0 realization in Cayuga. This study's corpus consisted of approximately 52 min of spoken Cayuga narratives by two L1 speakers (2736 hand-segmented word tokens). The results of the study corroborated some of the earlier descriptions. Words for both speakers exhibited prominence on word-finally in utterance non-final contexts. In utterance final contexts, however, sensitivity to word shape, differed between speakers. By comparing the two locations of prominence, we note small differences both in the alignment and scaling of peaks. Word-final prominence is, on average, scaled higher than utterance-final prominence (210 versus 204 Hz). Finally, utterance-final prominence is aligned earlier in the syllable than word-final prominence (33% vs 49% of vowel duration respectively).

**4aSCb2. Uvular rhotic weakening in Yiddish adjectival suffixes.** Guy Tabachnick (Linguist, New York Univ., 10 Washington Pl., New York, NY 10003, guyt@nyu.edu)

In traditional Yiddish dialects, the presence versus absence of word-final rhotics after unstressed [ɜ] in adjectival suffixes carries a heavy functional load, making distinctions of gender, number, and case. Belk *et al.* (2019) note that some Yiddish speakers with uvular rhotics do not fully articulate them word-finally, endangering this crucial distinction and perhaps contributing to the loss of gender and case in the Yiddish of contemporary Hasidic communities. This study analyzes adjectival endings in publicly available recordings of four speakers with uvular rhotics. The majority of speakers generally do not produce an audible [ʀ] or [ʁ] before consonants, but for these speakers, an underlying rhotic conditions lower F1 and sometimes higher F2 in the pre-rhotic vowel [ɜ]. These formant differences tend to be relatively consistent over the course of the vowel, suggesting that they are phonologized, rather than a phonetic effect due to coarticulation with a following uvular rhotic whose articulation may be covert. Word-final [ɜ] similarly exhibits lowered F1 and higher F2 before a word starting with [ʀ] or a dorsal fricative [x/ü], suggesting that coarticulation with a following dorsal is the phonetic precursor to the phonological effect observed for underlying rhotics.

**4aSCb3. Syllable shape effects on temporal aspects of the Danish stød.** Jailyn M. Pena (Linguist, New York Univ., 3 E. 124th St., Unit 2, New York City, NY 10035, jmp987@nyu.edu)

Danish phonologically contrasts modal phonation with "stød," a type of nonmodal phonation related to creaky voice (Fischer-Jørgensen, 1989). Stød (?) is licensed in a syllable by either a long vowel or a vowel + sonorant sequence (**CV:?**(**O**bstruent/**S**onorant), **CVS**?). Previous research noting high inter- and intra-speaker variability in stød realization has led researchers to claim that temporal aspects of stød production are random and not under speakers' control (Grønnum & Basbøll, 2007). In contrast, this study provides novel evidence that, overall, stød onset is timed with respect to syllables' rime midpoint and stød offset with respect to sonorant rime offset in monosyllabic words, though differences based on rime content persist across word types. 10 native Danish speakers were recorded reading sentences with embedded monosyllabic, stød-bearing words (CV:?(O/S), CVS?), and stød durations were marked. Comparison of different measures of stød onset (lag from either rime onset, rime midpoint, or vowel midpoint) revealed that lag from rime midpoint minimized variance in stød onset timing across word types. Furthermore, CV:?O/S words had earlier onset/offset times than CVS? words, which had earlier onset times than CV:? words. Together, these results indicate that stød production across different word types is not random but in fact modulated by syllable shape.

**4aSCb4. Initial weakening in Mixtecan languages.** Christian DiCanio (Dept. of Linguist., Univ. at Buffalo, Buffalo, NY 14260, cdicanio@buffalo.edu) and Jared Sharp (Linguist., Univ. at Buffalo, Buffalo, NY)

Prosodic boundaries influence patterns of consonantal strengthening and weakening across languages (Fougeron and Keating, 1997; Kakadelis, 2018; Katz and Fricke, 2018; Keating *et al*., 2003; White *et al*., 2020). Onset obstruents/nasals in prosodically prominent positions are lengthened and produced with greater contact between active and passive articulators (Fougeron and Keating, 1997; Fletcher, 2010; Keating *et al*., 2003; Lavoie, 2001). Typically, articulatory strengthening occurs in utterance-initial and word-initial position while weakening/reduction occurs in word-medial position (Katz and Fricke, 2018). We provide phonetic evidence that certain obstruents in *word-medial,* pre-tonic position are both lengthened and strengthened in Itunyoso Triqui, an indigenous language of Mexico, while word-initial obstruents are often shorter and reduced. We examined 67 min (8933 segments) from a spontaneous speech corpus produced by nine native speakers; and measured duration, voicing lenition, and spirantization. Onset consonants were lengthened slightly in stem-final (stressed) syllables but shortened elsewhere. For sonorant consonants, no utterance-initial lengthening was observed. These findings agree with previous work on a related Mixtecan language—Yoloxóchitl Mixtec (DiCanio *et al*., submitted)—and pose unique challenges for views hypothesizing that consonantal strengthening is either (a) universal or (b) serves to enhance word-level parsing (Katz and Fricke, 2018; White *et al*., 2020).

**4aSCb5. The acoustic realization and timing of glottal stops in Hawaiian.** Lisa Davidson (Linguist., New York Univ., 10 Washington Pl., New York, NY 10003, lisa.davidson@nyu.edu)

Studies of glottal stop realization have shown that it is often not produced with full closure (Ladefoged and Maddieson, 1996), but conditioning factors are not well understood. This study focuses on Hawaiian, which has phonemic glottal stop that is contrastive in both word-initial (/ʔaka/ "laugh," [aka] "shadow") and word-medial position (/pua/ "flower," /puʔa/ "to excrete"). Glottal stop realization is examined with respect to word position, different versus identical flanking vowel (/puʔu/ "hill"), and duration of the target /VʔV/ sequence. Data from eight native speakers come from the Ka Leo Hawaiò£i radio program. The majority of glottal stops are produced as a period of creaky voice (63%), either in a modal-creaky-modal, modal-creaky or creaky-modal configuration. Full closures (6.6%) were more likely in word-initial position, and identical flanking vowels led to longer periods of creak. Some tokens had only an intensity dip. Shorter target intervals had longer proportions of creak. These findings for the glottal stop phoneme are consistent with research on the timing of contrastive voice quality in vowels, which show a preference for modal-nonmodal-modal patterns to ensure that vowel quality, voice quality, and tone are recoverable (Silverman, 1997). Effects of word position and flanking vowel are also related to recoverability and segmentation.

**4aSCb6. Intonation processing in American English is incremental and supports forward-backward inference.** Jennifer Cole (Linguist., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208-0854, jennifer.cole1@northwestern.edu), Timo Roettger (Inst. for Cognit. Sci., Univ. of Osnabrück, Osnabrück, Germany), and Daniel Turner (Linguist, Northwestern Univ., Evanston, IL)

We test the hypothesis of incremental processing of intonation in American English, which predicts that pitch accent cues in the speech signal are immediately processed for inferring the referential status (focus, givenness) of the current word. A challenge to this model is the fact that a word is perceived as having pitch prominence only in relation to nearby words with lesser prominence. We test the time course of processing for pitch accents in American English in a speech comprehension experiment using eye-tracking. 34 native English speakers listened to instructions to click on one of four pictures (potential referents) displayed on the computer screen. Lexical content alone was not sufficient to identify the correct picture from the verbal instruction, but the accent pattern over the subject and object noun provided decisive cues. Findings from eye-tracking data show listeners use accent as a cue to referential meaning. An accented subject noun triggers immediate local inferences about the noun's referent (in looks to that picture) and forward inferences about the referent of the downstream object noun. Backward inferences (looks to the subject referent) are triggered by an unaccented object. These findings show incremental processing in both forward and backward inference for referential meaning.

**4aSCb7. Phrase-final lengthening and phonological vowel length in Lebanese Arabic.** Niamh Kelly (English Dept.American Univ. of Beirut, P.O.Box 11-0236, Beirut 1107 2020, Lebanon, nk114@aub.edu.lb)

The effect of phrase position on duration of words and segments, and its interaction with phonological vowel length, was examined in Lebanese Arabic. All words were disyllabic with initial stress, and the stressed syllable had either a phonologically long or short vowel: 'CV(V)(C).CV(C). Words were produced in carrier sentences, where one was sentence-medial and one sentence-final, both under contrastive focus so as to control for that effect. The measures examined were durations of the word, onset consonants, stressed and unstressed vowels, and intervocalic consonants. Six speakers produced 475 tokens. Speaker, token, vowel and syllable structure were included as random factors in the linear regression models. Results showed that unstressed vowels were longer in phrase-final position, due to final lengthening, while onsets were longer in phrase-medial position. Stressed vowel duration was longer in phrase-final position but only when the vowel was phonologically long. Neither word nor intervocalic consonant durations were affected by vowel length or phrase position. These findings show an asymmetric effect of phrase position based on vowel length, as well as an overall balancing of word duration by compensation between onsets and unstressed vowels. This research contributes a description of phrase-level effects on segmental duration patterns in this variety.

**4aSCb8. Reducing phonetically incomplete application of tone three sandhi to articulatory implementation.** Chin-Ting Liu (Dept. of Appl. English, National Chin-Yi Univ. of Technol., No.57, Sec. 2, Zhongshan Rd., Taiping Dist., Taichung 41170, Taiwan, ctjimboliu@gmail.com)

Mandarin Chinese Tone Three Sandhi (T3S) is a phenomenon where the first Tone 3 (T3, a falling-rising tone) syllable becomes a rising pitch when it is followed by another T3 syllable. The purpose of this study is to examine if the phonetically incomplete application of T3S among pseudowords could be reduced to the differences in articulatory implementation triggered by lexical familiarity effects. Four types of disyllabic T3 words were used, including two actual occurring morphemes (AO-AO), AO with an accidentally gapped morpheme (AO-AG), AG-AO, and AG-AG. Four acoustical parameters, including average fundamental frequency (f0), f0 contour, the turning percentage (lowest point) of the f0 contour and the f0 slope, were measured. Production results from thirty adult native speakers (with balanced gender) indicated that the application of T3S was phonetically incomplete for AG-AO and AG-AG word groups, showing that the morpheme type of the first syllable was the critical factor. Together with previous studies (Hsieh, 1970, 1975; Wang, 1993; Chuang *et al.*, 2011; Wee, 2019), this study supported Zhang and Peng's (2013) view that the covert, phonetic contrasts observed between real words and certain pseudoword types can be attributed to differences in the articulatory implementation triggered by lexical familiarity effects.

**4aSCb9. Right-edge markers of narrow focus in Japanese: The duration and intensity of the genitive particle "-no."** Marta Ortega-Llebaria (Linguist., Univ. of Pittsburgh, 4200 Fifth Ave., Pittsburgh, PA 15260, m.ortega.llebaria@gmail.com) and Jun Nagao (Foreign Lang., Gifu Shotoku Gakuen Univ., Gifu-shi, Gifu, Japan)

Because duration and morae express lexical contrasts in Japanese, e.g., ojiisan "old man" vs ojisan "uncle," narrow focus (e.g., I prefer the WHITE horse) is expressed mainly by pitch cues. Yet, in colloquial registers, adding a mora to some qualifiers conveys emphasis (e.g., *sug**o**ku tsukareta* "very tired"—*sug**oo**ku tsukareta* "VERY tired"). We investigated whether duration and intensity played a more important role in marking narrow focus than previously assumed in the literature by examining the duration and intensity of the genitive particle [-no] in 3 contexts: (1) attached to a focused word (*UMÁNO hizume* "HORSE's hoof"), (2) pre-focal position (*umáno HIZUME*), and (3) neutral statements (*umáno hizume*). 105 Noun-no Noun combinations in these contexts were elicited from 6 native speakers of Japanese. Results showed that duration cued narrow focus in context (1) by lengthening the whole word (UMANO) and by lengthening [-no] with respect to the root as if adding a mora like in *sugooku*. Intensity, however, marked the end of an Accentual Phrase independently of its focus. To gain a better understanding of the Japanese narrow focus mechanism, we are currently labeling pitch cues in our database in order to examine their interaction with duration.

THURSDAY MORNING, 10 DECEMBER 2020 — 11:15 A.M. TO 12:00 NOON

## Session 4aSCc

## Speech Communication: Prosody II (Poster Session)

Authors will be at their posters from 11:15 a.m. to 12:00 noon.

### *Contributed Papers*

**4aSCc1. An acoustic study of the contrast between clear and plain speaking style for Mandarin.** Paul Tupper (Mathematics, Simon Fraser Univ., Burnaby, BC, Canada, pft3@math.sfu.ca), Keith Leung, Yue Wang (Linguist., Simon Fraser Univ., Burnaby, BC, Canada), Allard Jongman, and Joan Sereno (Linguist, Univ. of Kansas, Kansas City, KS)

This study examines the acoustic characteristics of clear and plain conversational productions of Mandarin tones. Twenty-one native Mandarin speakers were asked to produce a selection of Mandarin words in both plain and clear speaking styles. Several tokens were gathered for each of the four tones giving a total 2045 productions. Seven critical tonal cues were computed for each production: fundamental frequency (F0) mean, slope, curvature (computed by a parabolic fit to the tone contours), duration, mean intensity, location of maximum intensity, and a binary variable coding whether the production involved creaky voice. A linear mixed-effects regression model was used to explore how these cues changed with respect to the clear versus plain distinction, with fixed effects being speaking style and tone and the only random effect being speaker. The strongest effects detected were that duration and mean intensity increased in clear speech across speakers and tones. Other cues did not show consistent changes with speaking style. An additional finding was that, for contour tones, speakers accomplished the increase in duration by stretching out the tone contours in time while not changing the F0 range. These results are discussed in terms of universal versus tone-specific clear speech modifications.

**4aSCc2. Interactions between fundamental frequency and fricative spectral balance in Mandarin Chinese: A corpus study.** Heather Johnson (Linguist., Boston Univ., 621 Commonwealth Ave., Boston, MA 02215, heathdj@bu.edu) and Jonathan Barnes (Linguist., Boston Univ., Boston, MA)

Voiceless segments create obvious problems for the realization of linguistic tone contours. Niebuhr 2012 hypothesizes that speakers compensate for gaps in F0 using non-F0 cues to support threatened contrasts. For example, in several intonation languages, spectral balance in voiceless fricatives appears to covary in certain contexts with local F0 levels, suggesting that listeners may use raised spectral center-of-gravity (CoG) as a perceptual proxy for high F0 where the latter is absent. To our knowledge, however, this pattern has yet to be identified in a lexical tone language. This study investigates Mandarin Chinese voiceless fricative spectra in contrasting lexical tone contexts, using data from the AISHELL-1 Mandarin Speech Corpus (one talker, 3,562 words). 778 fricatives were located using the Penn Phonetics Forced Aligner for Chinese, and measurements for closure duration, intensity, CoG, and surrounding F0 levels were taken with Praat. Both consonant place and following vowel context exhibited expected effects on fricative CoG. Preceding and following lexical tone categories did not. We did, however, identify a small but significant positive relationship between following F0 levels, and fricative CoG. This is discussed both in terms of compensation for missing F0, and as an indirect consequence of increased intensity under raised F0.

**4aSCc3. Complex interplay of lexical stress and focus prominence in Tohono O'odham.** Daejin Kim (Linguist., Univ. of New Mexico, 1 University of New Mexico, MSC03 2130, Albuquerque, NM 87131-0001, daejinkim@unm.edu) and Robert Cruz (Linguist., Univ. of New Mexico, Albuquerque, NM)

This paper explores how the linguistic contrast (focus prominence) is realized with the rhythmic and timing pattern, evidenced in the typologically rare and complex plural noun reduplication (CVC → CVCVC; e.g., *ban* "a coyote" versus *baban* "coyotes") in Tohono O'odham (TO). Our preliminary result of two native speakers of TO indicates that the greater focus effects (lexically and morphologically contrastive focus) are realized with the *higher f0* at the first CV sequence and the *lower f0* at the second CV sequence, strengthening the language-specific strong-weak (SW) stress pattern. However, no focus effect was found on the durational properties. Also, no significant morphologically contrastive focus effect, contrasting singular and plural forms of the target words, was found on both *f0* and durational properties. Our finding implies that the realization of prosodic and phonological elements in Tohono O'odham is also conditioned with fine-grained phonetic details. We also discuss the future possibility of the complex phonetics-prosody interplay of the language.

**4aSCc4. A syntactic-prosodic analysis of focalizing "ser."** Tanya Flores (Lang. and Lit., Univ. of Utah, 255 S Central Campus Dr., LNCO 1400, Salt Lake City, UT 84109, Tanya.Flores@utah.edu) and Catalina Mendez Vallejo (Princeton Univ., Princeton, NJ)

This project examines the relationship between an innovative syntactic structure in Colombian Spanish and its prosodic description. The focalizing *ser* (FS) is a dialectally marked syntactic structure in which *ser* is generated in a clause-internal focus phrase and functions as a discourse link between given and new information (Méndez Vallejo, 2009; Pato, 2010). The following is an example of FS for "It was the puppy who left barking": *Salió ladrando fue el perrito* leave-3SG-PAST bark-PROGR be-3SG-PAST the puppy This is the first analysis on how FS is marked prosodically, and how it compares to cleft and pseudo-cleft focus structures. Data were collected from 40 speakers, 10 from each of four Colombian cities: Barranquilla, Bogotá, Cali, and Medellín. Our results show that FS behaves differently from other focus structures in terms of prosodic marking. Unlike *ser* in cleft and pseudo-cleft structures, *ser* in FS structures can be prosodically marked with a prenuclear pitch accent. This study reveals that although FS is considered a discourse or emphatic marker (Curnow and Travis, 2004; Pato, 2010), it can behave like a content word in terms of syntax and prosody, even receiving prosodic focus marking.

**4aSCc5. Prosodic encoding of information structure in nuclear and prenuclear positions in American English.** Jennifer Cole (Linguist., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208-0854, jennifer.cole1@northwestern.edu) and Eleanor Chodroff (Lang. and Linguistic Sci., Univ. of York, York, United Kingdom)

A hallmark of English intonation is the prosodic encoding of information structure (givenness, focus) through distinctions in acoustic prominence and pitch accenting. Information structure (IS) effects are typically illustrated for nuclear accents (rightmost), while less is known about prenuclear accents. We report on a production study of American English, comparing acoustic correlates of accentual prominence (Duration, Intensity, F0) for words as a function of position (Prenuclear, Nuclear), and IS (Contrastive, New, Accessible, Given). 64 native English speakers read aloud 20 ministories ending with a target SVO sentence. In a between-subjects manipulation, the IS of the target subject OR the target object varied across stories (Contrastive, New, Accessible, Given). Results show main effects of Position (nuclear words have longer duration and lower intensity than prenuclear), and IS (given words have shorter duration and lower intensity; new words are longer). Interaction effects indicate that positional asymmetries persist when focus conditions are matched (a contrastive nucleus has greater prominence than a contrastive prenuclear word), and Position and IS effects are additive when they are in the same direction. Overall, we observe robust positional effects on acoustic prominence, and weaker, partial effects of IS, similar in both positions. Prosody primarily encodes position, secondarily IS.

**4aSCc6. Prosodic structural and tonal contextual modulation of Voice Onset Time and consonant-induced fundamental frequency in the three-way laryngeal contrast in Thai.** Alif Silpachai (Appl. Linguist. and Technol., Iowa State Univ., 527 Farmhouse Ln., Ames, IA 50011-1054, alif@iastate.edu)

Voice Onset Time (VOT) and consonant-induced fundamental frequency (CF0) may signal phonological laryngeal contrasts. This signaling may be modulated by prosodic structure and pitch context. However, such modulation in tonal languages is unclear. This study investigates the roles of prosodic structure and tonal context in the phonetic implementation of laryngeal contrasts in Thai, a tonal language with a three-way laryngeal contrast. Monosyllabic words with /b/, /p/, or /p$^h$/ as the onset, bearing the falling (51), mid (32), or low (21) tone, and produced at the Intonational Phrase (IP) boundary or the Word boundary were analyzed. Results showed modulation of VOT and CF0 by prosodic structure and tonal context across laryngeal contrasts. The VOTs of /b/ and /p$^h$/ at the IP boundary were longer compared to their counterparts at the Word boundary. The CF0s of /b/ and /p/ across all tones were lower at the IP boundary compared to their counterparts at the Word boundary. An observed CF0 effect was more visible with the falling tone compared to the mid tone; no such effect was found with the low tone. Within the falling and mid tones, the CF0 effect occurred more at the IP boundary compared to at the Word boundary.

**4aSCc7. Global and local pitch level in Caabe is not predicted by the frequency code.** Tajudeen Mamadou Yacoubou (Linguist, Rutgers Univ., 18 Seminary Pl, New Brunswick, NJ 08901, dine.mamadou@rutgers.edu), Mariapaola D'Imperio, Huteng Dai, and David Kleinschmidt (Psych., Rutgers Univ., New Brunswick, NJ)

The tendency for questions to be realized with a high or rising pitch has long been held as a (near-) universal property of world languages (Bolinger, 1978; Ohala, 1984, a.o) and even lead to ethological theories as the "Biological Codes," in particular the "Frequency Code" (Gussenhoven, 2002). Rialland (2007,2009) documented a variety of languages found along the Sudanic Belt in Africa, which realize their questions with a terminal non-high pitch (aka "Lax Prosody"), characterized by longer prefinal duration. Also, in a study of five Ghanaian languages, Cahill (2013) found that the overall pitch level in statements is higher than that of questions. In the present paper, we present findings from a production study involving 24 native speakers of Ede Chaabe (Niger-Congo) to answer the questions: (1) do global and local f0 levels differ as a function of modality? and if so, (2) in which direction? Preliminary results show that questions in Ede Chaabe, contra the Frequency Code hypothesis, present both a local final fall as well as a significantly lower overall pitch level compared to statements. Furthermore, tonal specification interacted with intonation specification, in that L tone utterances did not show a significant level difference induced by modality.

**4aSCc8. Phonetic evidence for categorical differences in prosodic structure.** Seung-Eun Kim (Linguist, Cornell Univ., 203 Morrill Hall, Ithaca, NY 14853, sk2996@cornell.edu) and Sam Tilsen (Linguist, Cornell Univ., Ithaca, NY)

An experiment was conducted to assess phonetic evidence for categorically distinct prosodic structures associated with two types of relative clauses in English. Non-restrictive relative clauses (NRRCs) and restrictive relative clauses (RRCs) have been argued to be typically produced with different prosodic phrase structures. To test whether there is evidence for this, productions of the two relative clauses were elicited. A wide range of variation in speech rate was elicited by using a moving visual analogue which cued participants for rate variation. Acoustic and articulatory data were collected from twelve participants. We assessed whether the functional relations between speech rate and various phonetic measures at phrase boundaries differed by syntactic context. In addition, linear and sigmoidal models were fit to each of the articulatory and acoustic measures within each syntactic context, and the corrected Akaike Information Criterion (AICc) was used to determine whether the sigmoidal model provides a substantially better fit than the linear model. Although most of the phonetic measures showed a significant difference between the two syntactic structures, which provides some evidence for distinct prosodic categories, the

non-linearity analyses in both structures showed weak evidence for categorical variation in prosodic structure.

**4aSCc9. The acoustics of word stress in Gaeilge Chorca Dhuibhne.** Anton Kukhto (Linguist & Philosophy, MIT, 77 Massachusetts Ave., Cambridge, MA 02135, kukhto@mit.edu)

Gaeilge Chorca Dhuibhne (GCD) is a variety of Irish spoken in the Dingle peninsula, Co. Kerry, Ireland. Word stress in GCD falls on the second syllable if it is heavy (i.e., contains a long vowel or a diphthong), on the third syllable if it is heavy and the preceding syllables are light, and on the first syllable in other cases. While formal phonological analyses of this system have been proposed before, details of the acoustics of stress remain poorly understood. This paper reports on a production study conducted with five native speakers of GCD in the villages of Ceann Trá and Baile an Fheirtéaraigh. A list of 64 words of various prosodic shapes in carrier-phrases was recorded. The stimuli were presented to the participants in written form and read out loud. The phrases were constructed in a manner that would allow to elicit target words in positions where they are versus are not associated with a HL phrasal accent in order to disentangle effects of word- and higher-level prosody. The preliminary results show that stressed syllables are characterised by a higher $f_0$ and higher intensity levels compared to unstressed syllables, whereas duration is not a significant correlate of stress.

**4aSCc10. A measure of pause frequency correlation with post traumatic stress disorder symptom severity.** Richard A. Southee (English, Arizona State Univ., 110 West Bell Rd., Apt. 276, Phoenix, AZ 85023, rsouthee@asu.edu)

Cognitive science research has found prosodic qualities of speech can be analyzed to establish development of cognitive and neurological disorders, as well as treatment efficacy, primarily for disorders such as Parkinson's Disease and Alzheimer's Disease. Linguistic research has largely overlooked prosodic features within research on Posttraumatic Stress Disorder and instead focused on semantic choices, despite PTSD's impact on cognitive function. The present pilot study aims to extend findings within cognitive sciences to research on PTSD to establish if prosodic qualities of speech can be used in a similar manner, specifically analyzing pause frequency. Student veterans from universities across the US were recruited for participation. Participants were prompted to elicit a spontaneous monologue between 90 and 120 s that was recorded, they were also required to fill out a PCL-5. Speech was analyzed for pauses of 0.25 s or greater via speech analysis software(PRAAT), and the frequency was contrasted against the individual PCL-5 scores. The results of the pilot study were inconclusive due to limitations in the participant pool. As such this highlights needs in further investigation, but establishes the methods used as functional. Future research should be focused on extending the procedures to a larger pool and looking at other prosodic features.

THURSDAY MORNING, 10 DECEMBER 2020 — 9:30 A.M. TO 11:00 A.M.

## Session 4aSPa

## Signal Processing in Acoustics, Underwater Acoustics, Computational Acoustics, Acoustical Oceanography, and Animal Bioacoustics: Machine Learning in Acoustics III

Erin M. Fischell, Cochair
*Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, 266 Woods Hole Rd., MS 11, Woods Hole, MA 02543*

Daniel Plotnick, Cochair
*Penn State, Penn State University, State College, PA 16804*

Wu-Jung Lee, Cochair
*Applied Physics Laboratory, University of Washington, 1013 NE 40th St., Seattle, WA 98105*

**Chair's Introduction—9:30**

### *Contributed Papers*

**9:35**

**4aSPa1. Explaining neural network predictions of acoustic fields in ideal single- and multi-path environments.** Brandon M. Lee (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109, leebm@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Neural networks are an increasingly popular tool for solving a wide variety of problems. Given some up-front computational effort, the complex patterns which exist in large datasets can be embedded in an explicit, though difficult to interpret, neural-network 'formula' for future predictions. As reliance on neural networks increases, so does the demand for their explainability, since practitioners must create safe and unbiased methods. In this presentation, an effort is made to better explain and understand neural networks which are trained to provide solutions to the point-source Helmholtz-equation in axisymmetric single-path, two-path, and multi-path (ideal waveguide) environments having constant sound speed. This analysis emphasizes source frequencies in the 100s of Hz, depths up to 500 m, and

ranges up to 1 km for sound speeds near 1500 m/s. In all cases, the neural networks' intermediate computational values are compared to those of the corresponding analytical Helmholtz-equation solution used to generate the neural networks' training dataset. The effects of the choice of neural network inputs, noise-like inputs, and correlated input features are also considered. The insights developed from this investigation provide insights for the extension of neural network predictions of acoustic fields to more complex environments. [Work supported by the NDSEG fellowship program.]

**9:55**

**4aSPa2. Using machine learning to evaluate the fidelity of acoustic simulations.** Andrew J. Miller (Brigham Young Univ., N283 ESC, Provo, UT 84602, ajm913@gmail.com), Scott D. Sommerfeldt, and Jonathan D. Blotter (Brigham Young Univ., Provo, UT)

There has been a longstanding tradeoff in evaluating the quality or fidelity of sound recordings: subjective listening tests are time consuming and expensive, but objective measures often fail to capture the nuances of human perception. The research presented here seeks to address this problem by investigating the use of machine learning to evaluate the fidelity of acoustic simulations. To begin, we created a dataset of recordings representing varying levels of audio fidelity. Participants listened to each of the recordings and subjectively classified the perceived fidelity. Various audio features were extracted from the recordings, including several psychoacoustic sound quality metrics and other features commonly used in speech recognition/assessment and music genre classification. These features were input to various machine learning algorithms to test which best modeled the human classifications. A logistic regression model was initially determined to be the most advantageous, dependent on using binary classification. Introducing a reference sound, and calculating each feature relative to the reference, significantly improved accuracy.

**10:15**

**4aSPa3. Differences in regression, classification, and multi-task deep learning on pressure time series for range and seabed type.** David F. Van Komen, Kira Howarth (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., N251 ESC, Provo, UT 84602, tbn@byu.edu), and David P. Knobles (Phys., Knobles Sci. and Anal., Austin, TX)

Estimating source-receiver range and predicting seabed type are important problems in underwater acoustics. Deep learning solutions for simultaneous predictions have been previously shown some success for these problems due to their ability to learn patterns from large datasets. However, one important question to consider in deep learning is how predictions should be made: A network can be configured to classify or regress predictions. In this study, that question is explored by comparing predictions from networks trained on simulated SUS charge pressure time-series configured to either regress or classify range and seabed predictions from measured data taken with the IVAR system during the 2017 Seabed Characterization Experiment. To further this inquiry, networks configured to regress range and classify seabed type (via "multi-task" learning) are also explored. Separating the two predictions proves to be useful, as the networks using multi-task learning perform better at predicting range and seabed class simultaneously than those configured only for classification or regression. The results of this experiment illustrate the need to use the proper type of network outputs depending on the desired predictions. [Work supported by the Office of Naval Research.]

**10:35**

**4aSPa4. Geoacoustic parameter estimation using machine learning techniques.** Gopu R. Potty (Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882, potty@egr.uri.edu), James H. Miller (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Vishakh R. Gopu (Viome, Inc., New York, NY), Zoi-Heleni Michalopoulou (New Jersey Inst. of Technol., Newark, NJ), and Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA)

The Airy phase region corresponding the minimum group velocity of the dispersive acoustic normal modes in shallow water are extremely sensitive to bottom properties. The group speed minima and the associated frequency data for two lower order modes were used to train a neural network to predict the bottom parameters in a previous study (Potty *et al.*, 2019). The geoacoustic model for the bottom consisted of a sediment layer over acoustic basement. Synthetic data generated by varying the values of the sound speeds in the sediment layer and basement were used to train and evaluate the technique. The output of the neural network was used as the background model for a linear perturbation inversion. Data collected from the Shelf break Primer experiment were used to test the algorithm and the preliminary results were promising. This study will continue the previous work by incorporating higher order modes in the training data, testing with data from other experiments and using other machine learning techniques. The performance of this hybrid approach (neural network combined with linearized inversion) will be compared with fully non-linear inversion, both in terms of accuracy and computation time. [Work supported by the Office of Naval Research.]

# Session 4aSPb

## Signal Processing in Acoustics, Underwater Acoustics, Computational Acoustics, Acoustical Oceanography, and Animal Bioacoustics: Machine Learning in Acoustics IV

Erin M. Fischell, Cochair
*Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, 266 Woods Hole Rd., MS 11, Woods Hole, MA 02543*

Daniel Plotnick, Cochair
*Penn State, Penn State University, State College, PA 16804*

Wu-Jung Lee, Cochair
*Applied Physics Laboratory, University of Washington, 1013 NE 40th St., Seattle, WA 98105*

**Chair's Introduction—11:15**

### *Invited Paper*

**11:20**

**4aSPb1. A brain for a batbot: Combining deep learning and biomimetic robots to understand and replicate bat biosonar.** Rolf Müller (Mech. Eng., Virginia Tech, 1075 Life Sci. Cir, Blacksburg, VA 24061, rolf.mueller@vt.edu), Xiaoyan Yin, Ruihao Wang (Mech. Eng., Virginia Tech, Blacksburg, VA), Liujun Zhang (Elec. and Comput. Eng., Virginia Tech, Blacksburg, VA), and Michael Goldsworthy (Comput. Sci., Virginia Tech, Blacksburg, VA)

Sonar-based navigation in complex natural environments—whether underwater or in-air—poses major scientific and technological challenges. A key factor for these problems is the unpredictable nature of echoes that are superpositions of contributions from many reflectors ("clutter"). However, many bat species thrive in dense vegetation and hence demonstrate every night that such echoes can convey copious amounts of sensory information. Deep learning (DL) provides a fresh look at these difficult problems with the possibility of taking performance to new levels. A critical advantage of DL is its superior ability to discover informative features. However, DL requires training data sets that are typically much larger than what can be obtained in experiments with behaving animals. Biomimetic robots that can reproduce the behaviors of bats are a good way to obtain large-enough data sets of real-world echoes that have been recorded under controlled conditions. Examples for this approach are the exploration of biosonar landmarks and passageway finding in natural environments. For even larger data sets with better control over the underlying parameters, data augmentation methods based on generative networks can be used. An important remaining challenges is understanding the features selected by the DL networks at the signal and physical levels.

### *Contributed Papers*

**11:40**

**4aSPb2. Active listening and learning for orca sound detection.** Kunal B. Mehta (Comput. Eng., K.J Somaiya Inst. of Eng. and Information Technol., Somaiya Ayurvihar Complex Eastern Express Hwy. Near Everard Nagar, Sion East, Mumbai, Maharashtra 400022, India, kunal07@somaiya.edu), Jorge Rodriguez Saltijeral (Tecnológico de Monterrey, Saltillo, Mexico), Jesse Lopez (Axiom Data Sci., Portland, OR), Abhishek Singh (National Inst. of Technol. Durgapur, Port Blair, India), Valentina Staneva (Univ. of Washington, Seattle, WA), Scott Veirs (Beam Reach (SPC), Seattle, WA), and Val Veirs (Beam Reach (SPC), Friday Harbor, WA)

Southern Resident Killer Whales are in danger of extinction, however, the current approaches to detect and study their habitat are very time and labor consuming. To streamline this process we have built an active learning system leveraging the vast streams of passive acoustic data collected by hydrophones, the advances in machine learning for audio analysis, and the expertise of trained bioacousticians. The system visualizes predictions of a machine learning algorithm on raw data and allows input from the user to verify and correct them, thus supplementing the existing training dataset and also engaging the user in the annotation process. Researchers can focus on labeling only observations for which the algorithm has high uncertainty, while most of the data get labeled automatically. We show that the active learning system improves the performance of a Convolutional Neural Network algorithm on a two-category classification problem (presence/absence of a call) even with very few extra annotations: the f1-score increased from .83 to .84 with 50 new annotations ($\sim$3% increase of the labeled dataset). We have designed the system flexible to incorporate other algorithms and facilitate result comparison within the community. It is also open source and easy to deploy on diverse computing infrastructures.

12:00

**4aSPb3. Unsupervised clustering of coral reef fish calls.** Emma Reeves Ozanich (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9525 Genesee Ave., Apt. 204, San Diego, CA 92121, ecreeves@ucsd.edu), Aaron M. Thode, Peter Gerstoft (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), Simon E. Freeman, and Lauren A. Freeman (Naval Undersea Warfare Ctr., Middletown, RI)

An unsupervised process is proposed for clustering and identifying fish calls in an acoustically active coral reef soundscape. First, potentially localizable acoustic events were detected on three directional autonomous seafloor acoustic recorders (DASARs), sampled at 1 kHz, using an automatic directional detector. A maximum likelihood localization algorithm was used to remove unlocalizable events. For each localizable event, standard acoustic metrics were extracted from the spectrogram and timeseries, while user-agnostic latent features were extracted from the spectrograms by an undercomplete convolutional autoencoder (CAE) neural network. Unsupervised clustering methods, including K-means and agglomerative hierarchical, identified distinct acoustic classes from both feature sets. The unsupervised clustering process was used to analyze data collected near a Hawaiian coral reef in February 2020. During a 24-h period, about 1 event per second was detected, with diel variation in the number of detected events. Compared to a hand labeled test set, the unsupervised clustering process with standard acoustic metrics was 91% accurate at identifying fish call trains, which were associated with dusk chorusing, as opposed to single pulse calls or Humpback whale sound. The tradeoffs between physical features and CAE latent features for unsupervised call discovery are discussed.

12:20

**4aSPb4. A deep learning approach to utilizing complex Doppler patterns for direction-finding inspired by bat biosonar.** Xiaoyan Yin (Mech. Eng., Virginia Tech, 1075 Life Sci. Cir, Blacksburg, VA 24061, xiaoyan6@vt.edu) and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Standard approaches for determining the direction of a sound source from pressure recordings require the use of multiple frequencies or multiple receivers. Inspired by the finding that certain bat species can move their ears fast enough to create Doppler-shift signatures, we have investigated a single pressure sensor that is surrounded by a moving baffle which mimics the mobile pinna of a bat. Due to its fairly complex geometry and motion patterns, the baffle created time-frequency signatures that were qualitatively observed to depend on target direction but could not be readily decoded. In order to investigate whether these patterns can be used for direction finding, we have devised a deep learning regression network with 18 convolutional layers to estimate target direction from a feature vector that represents the spectrogram of the Doppler signatures. The training data consisted of 25,620 signals associated with 1281 different directions (61 in azimuth and 21 in elevation). The regression network has been tested by coupling it with a biomimetic pinna that was mounted on a pan-tilt unit so that it could track an physical sound source (ultrasonic loudspeaker). The accuracy of this system was found to better than 1 deg (rms error).

THURSDAY MORNING, 10 DECEMBER 2020 — 9:30 A.M. TO 11:00 A.M.

# Session 4aUWa

## Underwater Acoustics: Underwater Acoustic Inversions

Derek Olson, Chair

*Oceanography, Naval Postgraduate School, Spanagel Hall, 833 Dyer Road, Monterey, CA 93943*

**Chair's Introduction—9:30**



### *Contributed Papers*

9:35

**4aUWa1. Normal modes separation by use of ambient noise cross-correlation function in shallow sea.** Sergei Sergeev (Acoust. Dept., Phys. Faculty, Moscow State Univ., Leninskie Gory, Moscow 119991, Russian Federation, sergeev@aesc.msu.ru), Ildar Sabirov, and Andrey Shurup (Phys. Faculty, Acoust. Dept., Moscow State Univ., Moscow, Russian Federation)

The possibility to extract the time-domain Green's function from ambient noise cross-correlation function was demonstrated by a number of authors. This approach has a weak point: the accumulation time of the noise signal. The accumulation time is significantly reduced when using mode signals and separating normal modes. Both numerical study of the spectrogram of the noise cross-correlation function and experiment in Arctic shelf demonstrated that dispersion curves observed in the spectrogram in the case of large distances between hydrophones, with a decrease in this distance are transformed into separate maxima localized both in frequency and in time delay. These maxima arise due to the presence of a stationary phase point near the minimum of the group velocities of the modes, and also due to modes interference, which is more efficient at small distances between hydrophones. At small distances, working with a signal in the frequency band where only the lowest hydroacoustic mode is excited has a number of advantages, the main of which is a decrease in the accumulation time. We demonstrate the change in the regime of modes separation with a change in the distance between hydrophones and discuss the optimal conditions for the applicability of the method.

9:55

**4aUWa2. Peculiarities of the Green's function reconstruction from the ambient noise cross-correlation function based on the mode approach.** Sergei Sergeev (Acoust. Dept., Phys. Faculty, Moscow State Univ., Leninskie Gory, Moscow 119991, Russian Federation, sergeev@aesc.msu.ru), Ildar Sabirov, and Andrey Shurup (Phys. Faculty, Acoust. Dept., Moscow State Univ., Moscow, Russian Federation)

It is known that the time derivative of the average ambient noise cross-correlation function is equal to the difference of the causal and anticausal time-domain Green's functions. This approach is often interpreted as "one

hydrophone acts as a source, the other as a receiver." Due to this interpretation and the principle of causality, the theory predicts the possibility of recovering only the imaginary part of the Green's function. In a real experiment, the correlation function is filtered and, as a consequence, information about its phase is saved. Our study, based on numerical calculations and experimental data, showed that it is possible to restore the phase of the Green's function. This phase experiences discontinuities at frequencies corresponding to the critical frequencies of the propagating hydroacoustic modes. This information is used to develop the new approach to reconstruct the shallow sea mode structure and can be used to reconstruct inhomogeneities in a passive tomographic scheme.

**10:15**

**4aUWa3. Construction of a Bayesian inversion method for determining seafloor sediment properties from simulated transmission loss data.** James Albritton (Mech. Eng., Univ. of Texas- Austin, 10000 Burnet Rd., Austin, TX 78758, jalbritton1050@gmail.com) and Aaron Gunderson (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Geoacoustic inversion of transmission loss measurements for determining sediment properties greatly reduces the time and cost needed to characterize the seafloor compared to cores or other *in situ* measurements. This study sought to determine the efficacy of a Bayesian inversion scheme to determine sediment parameters from transmission loss using simulated data, created using a hybrid finite element/ray acoustic model. For simplicity, the sediment was described as a fluid. A reduced order inversion parameter set was chosen to limit computational time and parameter intercorrelation, as well as to limit the search space to what is believed to be the most contributive parameters in the high frequency regime. This set includes sound speed, density, attenuation, and the von Karman spectral strength and exponent to describe interface roughness. The inversion scheme yielded marginal posterior probability distributions for each of the parameters, as well as information about parameter resolvability and covariance. [Work supported by the Office of Naval Research, Task Force Ocean.]

**10:35**

**4aUWa4. Seabed classification using localized forward modeling and deep learning.** Christina Frederick (Dept. of Mathematical Sci., New Jersey Inst. of Technol., Newark, NJ), Soledad Villar (Johns Hopkins, Johns Hopkins University, Whiting School, 3400 North Charles St., Baltimore, MD 21218, soledad.villar@jhu.edu), and Zoi-Heleni Michalopoulou (Dept. of Mathematical Sci., New Jersey Inst. of Technol., Newark, NJ)

We use full wave solutions to generate synthetic high-frequency data for training seafloor classifiers with acoustic signals in the ocean. We expect to recover high level details that cannot be achieved by standard forward models that do not include seafloor roughness. By creating acoustic templates using micro local analysis of full-wave simulations on smaller domains, we avoid the high cost of solving the equations on large domains, the main obstacle in full wave inversion. To demonstrate the usefulness of this modeling, we generate a training library of templates and show how well these can be used in classifying seafloor parameters. We consider two-layer seafloors with varying material types and thicknesses. We address the geoacoustic inversion problem with standard machine learning techniques and more sophisticated deep learning methods. The standard classifiers provide results with worse accuracy that do not generalize well to other test environments; the deep learning classifiers are more costly to train but have higher accuracy, generalizing better. We compare the performance with simulations performed at lower frequencies using a normal mode approach with no roughness effects, where the classification problem becomes simpler. [Work supported by NSF, EOARD, and ONR.]

THURSDAY MORNING, 10 DECEMBER 2020 — 11:15 A.M. TO 12:45 P.M.

## Session 4aUWb

## Underwater Acoustics: Scattering in the Ocean

Brian T. Hefner, Chair
*Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105*

**Chair's Introduction—11:15**

### *Contributed Papers*

**11:20**

**4aUWb1. Finite element model analysis of scattering by objects in complex seafloor environments.** Aaron M. Gunderson (Appl. Res. Labs., The Univ. of Texas at Austin, 10,000 Burnet Rd., Austin, TX 78758, aaron.gunderson01@gmail.com)

Underwater object detection and classification efforts are complicated by object burial within complex seafloor interfaces, which can alter the acoustic scattering signature in a variety of ways based on the level of object burial, orientation, frequency, grazing angle, and the variation of the seafloor topography. In this investigation, three-dimensional finite element models of acoustic scattering from objects in seafloor sediment were developed, to explore the effects that interface bathymetry and roughness have on the acoustic scattering, and to capture these effects within the model. Models were developed for comparison with measured acoustic scattering data from objects in real *in situ* ocean environments. Far-field scattering results outside the model domain were achievable through a numerical Green's function determination process. A high fidelity model that can account for the effects of interface rippling and mounding, small-scale roughness, and target/environment asymmetry has great appeal for detection and classification efforts. [Work supported by the Office of Naval Research and by the Strategic Environmental Research and Development Program.]

11:40

**4aUWb2. Inversion in rough surface scattering via the frequency-difference autoproduct.** Nicholas J. Joslyn (Appl. Phys., Univ. of Michigan, 450 Church St., Ann Arbor, MI 48109, njoslyn@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

The forward scattering of acoustic waves, with wavenumber $k$, from a rough surface is a well-known problem in underwater acoustics. A random rough surface, statistically characterized by its root-mean-square roughness $h$ and its correlation length $L$, exponentially reduces reflected field coherence as $kh$ increases. The frequency-difference autoproduct is a nonlinear mathematical construction that recovers field coherence in rough surface scattering by down shifting recorded signal frequencies, thereby minimizing apparent surface roughness. Salient features of frequency-difference-autoproduct forward-scattering results, including the dependence on both $h$ and $L$, are shown. Further, this presentation describes the unique capability afforded by the frequency-difference autoproduct to approach the inverse problem in a rough surface scattering environment. In particular, inference of a random rough surface's root-mean-square height and correlation length is accomplished via autoproduct-based signal processing methods. The work implements an isotropic surface roughness profile corresponding to in-band roughness values of $1.25 < kh < 3.5$. Using the same scattered signals, conventional in-band inversion techniques are compared to the results obtained via the frequency-difference autoproduct. Experimental results from a laboratory water tank may be discussed as well. [Work supported by ONR.]

12:00

**4aUWb3. Mid-frequency propagation and reverberation time series in a deep ice-covered ocean: Modeling and data analysis.** Anatoliy N. Ivakin (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, aniv@uw.edu) and Kevin L. Williams (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Mid-frequency (3.5 kHz) pulses (5 pings, 5 s-long, one per minute) were radiated by an omnidirectional source of opportunity in a deep (3 km) Arctic ocean at 30m depth under ice during the ICEX14 experiment, ALAS component [Williams *et al.*, *IEEE-JOE* **43**, 145–159 (2018)]. Data analysis and modeling results are presented for normalized (re source level) acoustic intensity received by an omnidirectional hydrophone located at a 719 m distance from and the same depth as the source. Received time series showed a clear direct blast signal followed by an about 30 s-long reverberation coda. Modeling of the under-ice propagation shows that magnitude of the direct signal is strongly affected by presence of a weak near-surface (within 50 m depths) acoustic channel and by reflectivity of the ice, controlled by its physical properties and acoustical parameters, particularly the ice layer thickness. Analysis of the reverberation coda shows effects of reflections and scattering from rough and/or heterogeneous bottom, as well as reflectivity of the ice layer. Some other effects that may be caused by variability of the sound speed profile and absorption in water are considered as well. Possibilities of inversions for water column, bottom and ice parameters based on this and similar experiments are discussed. [Work supported by ONR.]

12:20

**4aUWb4. Comparisons of integral equations and theoretical models of scattering from layered, rough seafloors.** Derek R. Olson (Oceanogr., Naval Postgrad. School, 833 Dyer Rd., 313b Spanagel Hall, Monterey, CA 93943, olson.derek.r@gmail.com)

Several models have been proposed for acoustic scattering from layered seafloors with rough interfaces. These include the small perturbation method, the Kirchhoff approximation, and three different small-slope approximations. In some situations, all of these models disagree. It is therefore advantageous to understand which model is appropriate to use in these circumstances. In this work, we use the integral equation to obtain exact, Monte-Carlo estimates of the reflection coefficient and scattering cross section. The exact results is compared to models for a single layer with a rough water-sediment interface, and a flat sediment-basement interface. It is found that the method of Jackson and Olson performs the best out of all models, although some of them agree when the rms roughness is small compared to the wavelength.

## Session 4pAAa

## Architectural Acoustics: Session in Honor of William J. Cavanaugh III

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

**Chair's Introduction—1:05**

### *Invited Papers*

**1:10**

**4pAAa1. Bill Cavanaugh–Reflections from abroad.** Joseph Soker (Top Consulting and Eng. Ltd., Jaffa 210, P.O.B 2835, Jerusalem 9102801, Israel, yossisoker1@topcons.co.il) and Alberto Haedo (Alberto M. Haedo, Ing., Buenos Aires, Argentina)

Bill Cavanaugh was enormously influential and well-known, with many international friends and colleagues. This presentation will offer some insights and stories about Bill as a friend and colleague to a couple of us, from Israel and Argentina. He encouraged our participation in professional societies, including the Acoustical Society of America (ASA) and the National Council of Acoustical Consultants (NCAC) which has become International, he offered much wisdom and advice, and he was a dear friend.

**1:30**

**4pAAa2. Bill Cavanaugh's contributions to the field of noise control engineering and INCE-USA.** Herb Singleton (Cross Spectrum Acoust., 25A Granby St., East Longmeadow, MA 01028, hsingleton@csacoustics.com) and Michael Bahtiarian (Acentech, Cambridge, MA)

This presentation will remember William J. Cavanaugh's contributions to the field of noise control engineering, and in particular his contributions to the Institute of Noise Control Engineering (INCE-USA). Bill served as INCE President in 1993, a board member, and an INCE Foundation board member for many years. He is a Fellow of INCE-USA; an honor given for his contributions to the advancement of noise control engineering as a teacher, author, consultant, researcher, and founding partner of a distinguished acoustical consulting firm. In 2015, Bill was honored again by INCE-USA with the award of the Laymon Miller Award for Excellence in Acoustical Consulting. This presentation will also summarize some of the ongoing INCE-USA activities, and its outlook for the future.

**1:50**

**4pAAa3. The National Council of Acoustical Consultants—A community held to high standards with gratitude to Bill Cavanaugh.** Noral D. Stewart (National Council of Acoust. Consultants, 7330 Chapel Hill Rd., Ste. 201, Raleigh, NC 27607, norals2020@sacnc.com), Eric Reuter (National Council of Acoust. Consultants, Portsmouth, NH), and Scott Pfeiffer (National Council of Acoust. Consultants, Chicago, IL)

The National Council of Acoustical Consultants (NCAC), founded in 1962, is an international organization of professional firms that specialize in acoustical consulting based upon education, references, and proven experience. All members must adhere to a strong Canon of Ethics supporting the highest standards of business practice, technical consulting, and client service. This presentation will quickly explore some of its history, discuss its recent activities supporting and promoting the profession, and share its outlook. Bill Cavanaugh was vital to its founding, growth, and prestige. His contributions will be discussed.

**2:10**

**4pAAa4. Bill Cavanaugh: The gold standard for acoustical consultants!.** Gary W. Siebein (Siebein Assoc.,Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com) and Keely M. Siebein (Siebein Assoc.,Inc., Gainesville, FL)

William (Bill) J. Cavanaugh was cited by the Acoustical Society for "*practical applications in building design and education in architectural acoustics and service to the Society*" when he received the Gold Medal. Bill set the "gold standard" for all people as well as for acoustical consultants for several generations. You can tell that he was trained as an Architect because of the joy that he found in working on challenging projects, the beauty of his hand sketches and solutions that resulted from his innovative and creative thinking and his desire to work with people through challenges. The field of architectural acoustics is perhaps more architectural because of Bill's contributions. His spearheading of his notable book with Joe Wilkes *Architectural Acoustics: Principles and Practice* was a major contribution to the field. His research, teaching and consulting work grew in parallel with the new discipline of architectural acoustics that he was shaping with his work. He was fundamental in beginning the Robert B. Newman Medal in architectural acoustics which has positively impacted the careers of many students. Above all, he was a man dedicated to family, faith and community in a way that serves as a shining example to us all.

THURSDAY AFTERNOON, 10 DECEMBER 2020 2:50 P.M. TO 4:20 P.M.

# Session 4pAAb

## Architectural Acoustics: Session in Honor of William J. Cavanaugh IV

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

**Chair's Introduction—2:50**

### *Invited Paper*

**2:55**

**4pAAb1. Bill Cavanaugh—Colleague, mentor, and friend.** K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

Bill Cavanaugh had a profound and positive influence on acoustics, on architectural acoustics consulting, and on many of us. This presentation will be a personal recollection through four decades with my mentor and friend Bill. This presentation will offer a few stories, will briefly review a few of Bill's areas of interest including cinemas and professional society activities, and will highlight several of his richly deserved awards, including the ASA Wallace Clement Sabine Medal in 2006, the NCAC (National Council of Acoustical Consultants) & INCE (Institute of Noise Control Engineering) Laymon N. Miller Award for Excellence in Acoustical Consulting in 2015, and the ASA Gold Medal in 2019.

THURSDAY AFTERNOON, 10 DECEMBER 2020 1:05 P.M. TO 2:15 P.M.

# Session 4pAB

## Animal Bioacoustics: Anthropogenic Sounds and Animal Bioacoustics

Derke R. Hughes, Chair
*NavSea Warfare Center, Division Newport, 1176 Howell St., Newport, RI 02841*

**Chair's Introduction—1:05**

4p THU. PM

### *Contributed Papers*

**1:10**

**4pAB1. Anthropogenic underwater sound emissions and other ecological effects on the marine environment.** Derke R. Hughes (NavSea Warfare Ctr., Div. Newport, 1176 Howell St., Newport, RI 02841, derke.hughes@navy.mil)

This presentation provides an overview of U. S. Navy and academia sponsored research to monitor, measure, model and mitigate the effects of active sonar sound transmissions on the health and well-being of marine mammals. Sound in the ocean has received considerable attention in the last several decades. Such intensive interest is in the sustainability and conservation of marine resources in ocean environments. Marine mammals in particular are sensitive to anthropogenic sounds. To capture the frequency-dependent nature of these sound effects on marine species, auditory computed weighting functions emphasize frequencies where a particular species is most susceptible to noise exposure. Researchers within academia and government continue development of functions like these and other metrics to characterize marine mammal responses to anthropogenic sound in the ocean. Underwater acousticians, marine biologists, oceanographers, engineers and computer scientists combine efforts to provide common standards of measuring and characterizing the anthropogenic sound effects on marine species. Further, computerized sonar equation models are used to: (1) provide standards of measuring and characterizing anthropogenic sound affecting various mammal species, and (2) determine metrics to guide policy makers' decisions in determining best practices for balancing marine environmental protection with national defense obligations.

**1:30**

**4pAB2. Cetaceans and seismic surveys in the Southern Adriatic Sea.** Ana Sirovic (Texas A&M Univ. Galveston, UCSD, 9500 Gilman Dr., MC 0205, La Jolla, CA 92093-0205, asirovic@ucsd.edu), Kait Frasier (Scripps Inst. of Oceanogr., La Jolla, CA), and Drasko Holcer (Croatian Natural History Museum, Zagreb, Croatia)

The first long-term, broadband, passive acoustic monitoring effort was conducted in the south Adriatic Sea from October 2018 to December 2019 using a High-frequency Acoustic Recording Package (HARP). The recording effort coincided with seismic surveys that took place off Montenegro in the fall and winter 2018–2019. These surveys occurred at distances 70 to 125 km from the recording site, at water depths between 70 and 800 m. Recordings were manually reviewed with start and end times of air-gun encounters and ship passages marked. A generic energy detector and unsupervised clustering procedure were used to extract odontocete echolocation clicks. During the surveys, sound pressure levels (SPL) in the 15 to 200 Hz band increased by up to 20 dB during fall and up to 10 dB during winter survey, with an average increase of 7.5 dB. Cuvier's beaked whale echolocation clicks were very commonly detected in the recordings, but sperm whale clicks were only sporadically present. At least three additional echolocation click types were also detected and their attribution to particular species is underway. Although mortality related to seismic surveys has not been recorded, high levels of noise have the potential to cause long term disruption to cetaceans in this region.

**1:50**

**4pAB3. Assessment of potential impacts to marine mammals from underwater radiated noise due to ferries.** Zachary Weiss (Noise Control Eng., LLC, 85 Rangeway Rd. Bldg. 2, Billerica, MA 01862, zfweiss@noise-control.com), William B. Bonnice, and Jesse Spence (Noise Control Eng., LLC, Billerica, MA)

Underwater radiated noise was measured and assessed from seven vessel classes active under the Washington State Department of Transportation Ferries Division (WSF). Unattended hydrophones were deployed in Seattle, Anacortes, and Port Townsend in order to collect data from nine vessels under multiple operating conditions. These data were then processed to identify potential Permanent Threshold Shift (PTS) and Temporary Threshold Shift (TTS) impacts to marine life for each WSF vessel class using the 2018 National Marine Fisheries Service (NMFS) guidelines. Behavioral Impacts (BI) were also evaluated according to criteria provided by WSF. Additionally, an assessment of the primary causes of radiated noise for each vessel class was performed and potential mitigation strategies were investigated.

THURSDAY AFTERNOON, 10 DECEMBER 2020 — 1:05 P.M. TO 1:35 P.M.

# Session 4pAO

## Acoustical Oceanography: General Topics in Acoustical Oceanography VII

John P. Ryan, Chair
*Research, MBARI, 7700 Sandholdt Rd., Moss Landing, CA 95039-9644*

**Chair's Introduction—1:05**

### *Contributed Paper*

**1:10**

**4pAO1. Quieting of low-frequency vessel noise in Monterey Bay National Marine Sanctuary during the COVID-19 pandemic.** John P. Ryan (Res., MBARI, 7700 Sandholdt Rd., Moss Landing, CA 95039-9644, ryjo@mbari.org), John E. Joseph, Tetyana Margolina (Naval Postgrad. School, Monterey, CA), Lindsey Peavey Reeves (National Marine Sanctuary Foundation, Silver Spring, MD), Leila Hatch (Stellwagen Bank National Marine Sanctuary, Scituate, MA), Andrew DeVogelaere (Monterey Bay National Marine Sanctuary, Monterey, CA), Brandon Southall (Univ. of California, Santa Cruz, CA), Alison Stimpert (Moss Landing Marine Labs., Moss Landing, CA), and Simone Baumann-Pickering (Univ. of California, San Diego, CA)

Low-frequency sound from large vessels is a major source of ocean noise, overlapping the range of marine animal communication. Changes in vessel activity provide opportunities to quantify the relationship between traffic levels and soundscape conditions in biologically important habitats. Using continuous deep-sea (890 m) recordings ~20 km from shipping lanes, we observed quieting of low-frequency noise within Monterey Bay National Marine Sanctuary (California, USA) during the COVID-19 pandemic. Noise levels in the frequency band 31–100 Hz, which captured large vessel noise while minimizing potential biases from geological and biological sound sources, decreased during January–June 2020. At monthly resolution the 2020 quieting trend correlated with decreasing cumulative hours of vessel presence derived from Automatic Identification System (AIS) data for cargo vessels ($r = 0.92$; $p = .01$), and the sum of all positively correlated vessel types (cargo, towing, dredging, passenger, tanker; $r = 0.88$; $p = 0.02$). February–June 2020 levels were up to 2.7 dB *re* 1 $\mu Pa^2/Hz$ below baseline of the two years prior. While anticipated to be short-term, at their peak these changes represent nearly a halving of acoustic power, emphasizing the significance of offshore large vessel traffic to sound levels in deeper waters in the sanctuary.

## Session 4pBAa

# Biomedical Acoustics and Signal Processing in Acoustics: New Developments in Lung Ultrasound III

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

Marie Muller, Cochair
*MAE, North Carolina State University, 911 Oval Drive, Engineering Building III, Raleigh, NC 27606*

**Chair's Introduction—1:05**

### *Contributed Paper*

**1:10**

**4pBAa1. A non-convex regularization based line artefact quantification method in lung ultrasound imagery for pulmonary disease evaluation.** Oktay Karakus (SCEEM, Univ. of Bristol, 1 Cathedral Square, VI Lab, Bristol BS1 5DD, United Kingdom, o.karakus@bristol.ac.uk), Nantheera Anantrasirichai (SCEEM, Univ. of Bristol, Bristol, United Kingdom), Adrian Basarab (IRIT, Univ. of Toulouse, Toulouse, France), and Alin Achim (SCEEM, Univ. of Bristol, Bristol, United Kingdom)

Lung (Pulmonary) diseases are among the most severe health problems which cause multiple deaths (more than 100 thousand people in the UK) every year. Moreover, following the COVID-19 pandemic, analysis and diagnosis of pulmonary disease became even more crucial. The common feature in all clinical conditions, both local to the lungs [e.g., pneumonia, chronic obstructive pulmonary disease (COPD)] and those manifesting themselves in the lungs (e.g., kidney disease, COVID-19) is the presence in lung ultrasound (LUS) images of a variety of artefacts. This work presents a novel method for line artefacts quantification in LUS images of pulmonary disease patients by using a non-convex regularization method. We employ a simple local maxima detection technique in the Radon transform domain, associated with known clinical definitions of line artefacts. Notwithstanding its non-convex characteristics, the proposed technique is guaranteed to converge through our proposed Cauchy proximal splitting (CPS) method and accurately identifies both horizontal (pleural, sub-pleural, A-) and vertical (B- and Z-) line artefacts in LUS images. The proposed method includes a two-stage validation mechanism, which is performed in both Radon and image domains to reduce the number of false and missed detections.

### *Invited Paper*

**1:30**

**4pBAa2. Automated segmentation and scoring of lung ultrasound images of COVID-19 patients.** Roshan Roshankhah (Mech. and Aerosp. Eng., North Carolina State Univ., 1840 Entrepreneur Dr., Raleigh, NC 27606, rroshan2@ncsu.edu), Yasamin Karbalaeisadegh (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC), Hastings Greer (Kitware, Inc., Clifton Park, NY), Federico Mento (Dept. of Information and Commun. Eng., Univ. of Trento, Trento, Italy), Gino Soldati (Azienda USL Toscana nord ovest Sede di Lucca, Lucca, Italy), Andrea Smargiassi, Riccardo Inchingolo (Dept. of Medical and Surgical Sci., Univ. Hospital Agostino Gemelli, Gemelli, Italy), Elena Torri (BresciaMed, Brescia, Italy), Stephen Aylward (Kitware, Inc., Clifton Park, NY), Libertario Demi (Information Eng. and Comput. Sci., Univ. of Trento, Trento, Italia, Italy), and Marie Muller (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC)

Using ultrasound for point-of-care lung assessment is becoming more and more relevant. Today, the outbreak of the coronavirus disease 2019 (COVID-19) has spread over the world at a very high rate. The most severe cases of COVID-19 are associated with lung damage such as ground-glass opacities and areas of lung consolidation, leading to acute respiratory distress. During the COVID-19 pandemic the need to detect and monitor the lung state is critical. Changes in the COVID-19 lung structure modify the way ultrasound propagates in the lung and are reflected by changes in the appearance of lung ultrasound images. Vertical artifacts known as B-lines appear and can evolve into white lung patterns in the more severe cases. Currently, these artifacts are assessed by trained physicians and sonographers, and the diagnosis is qualitative and operator dependent. We propose an automatic segmentation method using a convolutional neural network, to automatically stage the progression of the disease and predict the severity of the lung damage. By classifying the images based on illness severity we can define different scores—from healthy lung to most severe case—and produce a reliable tool to establish severity of COVID-19.

**1:50**

**4pBAa3. Assessment of COVID-19 in lung ultrasound by combining anatomy and sonographic artifacts using deep learning.** Shai Bagon (CS, Weizmann Inst. of Sci., 234 Hertzl St., Rehovot 7610001, Israel, shai.bagon@weizmann.ac.il), Meirav Galun, Oz Frank, Nir Schipper (CS, Weizmann Inst. of Sci., Rehovot, Israel), Mordehay Vaturi (Sackler Faculty of Medicine, Tel Aviv Univ., Tel Aviv, Israel), Gad Zalcberg (CS, Weizmann Inst. of Sci., Rehovot, Israel), Gino Soldati (Diagnostic and Interventional Ultrasound Unit, Valle del Serchio General Hospital, Lucca, Italy), Andrea Smargiassi, Riccardo Inchingolo (Dept. of Cardiovascular and Thoracic Sci., Fondazione Policlinico Universitario A. Gemelli IRCCS, Gemelli, Italy), Elena Torri (Bresciamed, Brescia, Italy), Tiziano Perrone (Policlinico San Matteo, Pavia, Italy), Federico Mento, Libertario Demi (Information Eng. and Comput. Sci., Univ. of Trento, Trento, Italia, Italy), and Yonina Eldar (CS, Weizmann Inst. of Sci., Rehovot, Israel)

When assessing severity of COVID19 from lung ultrasound (LUS) frames, both anatomical phenomena (e.g., the pleural line, presence of consolidations), as well as sonographic artifacts, such as A-lines and B-lines are of importance. While ultrasound devices aim to provide an accurate visualization of the anatomy, the orientation of the sonographic artifacts differ between probe types. This difference poses a challenge in designing a unified deep artificial neural network capable of handling all probe types. In this work we improve upon Roy *et al*. (2020): We train a simple deep neural network to assess the severity of COVID-19 from LUS data. To address the challenge of handling both linear and convex probes in a unified manner we employed two strategies: First, we augment the input frames of convex probes with a “rectified” version in which A-lines and B-lines assume a horizontal/vertical aspect close to that achieved with linear probes. Second, we explicitly inform the network on the presence of important anatomical features and artifacts. We use a known Radon-based method for detecting the pleural line and B-lines and feed the detected lines as inputs to the network. Preliminary experiments yielded $f1 = 68.7\%$ compared to $f1 = 65.1\%$ reported by Roy *et al*.

**2:10**

**4pBAa4. Detecting and localizing lung nodules by leveraging ultrasound multiple scattering.** Roshan Roshankhah (Mech. and Aerosp. Eng. Dept., North Carolina State Univ., 1840 Entrepreneur Dr., Raleigh, NC 27606, rroshan2@ncsu.edu), John Blackwell, Mir Ali, Behrooz Masuodi, Thomas M. Egan (Dept. of Surgery, Univ. of North Carolina, Chapel Hill, NC), and Marie Muller (Mech. and Aerosp. Eng. Dept., North Carolina State Univ., Raleigh, NC)

Although CT is widely used for detecting pulmonary nodules inside the parenchyma, the accurate real time lesion detection during video-assisted surgical procedures using ultrasound-based techniques is very attractive to improve to resection margins. Imaging the parenchyma using conventional B-mode is impossible due to multiple scattering in lung. However, the multiple scattering contribution of ultrasonic waves can be exploited, to detect the regions inside the lung with no scatterers. Nodules are homogeneous regions with relatively uniform properties compared to the healthy heterogeneous parenchyma containing millions of alveoli. We took advantage of this and developed an algorithm to extract multiple scattering contributions inside the highly scattering lung to localize pulmonary nodules. Inter-element response matrices were acquired using translated sections of a linear array transducer to semi-locally investigate the backscattered field. Extracting the multiple-scattering contribution using singular value decomposition and combining it with a depression detection algorithm allowed to detect regions with less multiple scattering, associated with the nodules. We validated this method in lung phantoms and demonstrated their feasibility in *ex vivo* pig and dog lungs containing artificial Vaseline nodules. We evaluated 7 lung blocks (4 pigs, 3 dogs) with multiple nodules and have localized all nodules in the last 4 lungs.

# Session 4pBAb

## Biomedical Acoustics and Signal Processing in Acoustics: New Developments in Lung Ultrasound IV

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

Marie Muller, Cochair
*MAE, North Carolina State University, 911 Oval Drive, Engineering Building III, Raleigh, NC 27606*

**Chair's Introduction—2:50**

### *Contributed Papers*

**2:55**

**4pBAb1. Using singular value distribution of backscattered ultrasound waves for tracking the pulmonary edema caused by COVID-19 in lung: A phantom-based study.** Omid Yousefian (Biomedical Eng., Columbia Univ., 630 West 168th St., Ste. VC 12-232B, New York, NY 10032, oy2138@columbia.edu), Julien Grondin, and Elisa Konofagou (Biomedical Eng., Columbia Univ., New York, NY)

A condition caused by COVID-19, pulmonary edema is an abnormal buildup of fluid in lungs. B-lines tracking of ultrasound images is a common way to track lung abnormality, however due to multiple scattering this method tracks the air volume change only on the lung surface. The aim of this phantom-based study is to assess the pulmonary edema in lungs using multiple scattering of ultrasonic waves in MHz range. A 128-element array transducer is used to form an Inter-elements Response Matrix (IRM) by firing a 5.2 MHz broadband pulse, element by element and receiving the backscattered signal by the whole array. The distribution of singular values of IRM in frequency domain is then analyzed for polymer sponges ($n = 4$) with different Water Volume Content (WVC) ranging from 5 to 25 ml mimicking pulmonary edema. The results of Random Matrix Theory are used to show that the mode of the Probability Density Function of singular values increases as the WVC increases in sponge indicating the change in scattering regime. The B-mode image of sponges with different WVC did not represent neither A nor B lines, however our results shows that there is a correlation between the WVC in sponge and the strongest singular value of IRM ($R = 0.89$) indicating the potential of using quantitative ultrasound in lung for early diagnosis of COVID-19.

**3:15**

**4pBAb2. Ultrasound characterization of lung mimicking phantoms using backscatter statistics.** Yasamin Karbalaeisadegh (North Carolina State Univ., 911 Oval Dr., Raleigh, NC 27606, ykarbal@ncsu.edu) and Marie Muller (MAE, North Carolina State Univ., Raleigh, NC)

The accumulation of fluids in lungs and subsequent reduction in air volume fraction (VF) is one of the main indicators of pulmonary edema. The backscattered ultrasound signals are affected by air VF resulting in a change in statistical parameters such as Shannon entropy which is defined as the negative of the logarithm of the probability distribution of the signal amplitude. This parameter has been exploited to characterize hepatic steatosis in the past. Two sponge phantoms are used to mimic lung parenchyma. To obtain 33 different air VFs (1.80%—65.82%), different amounts of water are added to each phantom; then, it is shaken to achieve a relatively homogeneous water distribution. A linear array transducer (128 elements) is used to transmit 5.2 MHz signals to the sample. The backscattered envelope of beamformed image data is acquired by a Verasonics Vantage system (Verasonics, Inc., Kirkland, WA). A sliding window method is used to calculate the entropy from local signals. The entropy is averaged over an region of interest (ROI) for samples with different air VFs. Results indicate a positive correlation between air VF and entropy ($R = 0.67$, $p \ll 0.001$) making entropy a potentially effective parameter to assess edema in lungs.

**3:35**

**4pBAb3. Ultrasound waves propagation in aerated inhomogeneous media.** Emanuele Peschiera (Information Eng. and Comput. Sci., Univ. of Trento, via Sommarive 9, Povo, Trento 38123, Italy, emanuele.peschiera@studenti.unitn.it), Thomas RIgolin (Information Eng. and Comput. Sci., Univ. of Trento, Trento, Italy), Federico Mento (Dept. of Information and Commun. Eng., Univ. of Trento, Trento, Italy), and Libertario Demi (Information Eng. and Comput. Sci., Univ. of Trento, Trento, Italia, Italy)

Lung ultrasound (LUS) has become a widely adopted diagnostic method for several lung diseases, recently also for COVID-19. However, LUS is mainly based on the presence of imaging artifacts, such as B-lines. These artifacts correlate with several pathologies, but their genesis is still partly unknown. Therefore, a study which focuses on the factors affecting these artifacts is carried out, numerically simulating through the MATLAB toolbox k-Wave ultrasound propagation inside the lungs. Since the main hypothesis behind the generation of B-lines is based on multiple scattering phenomena occurring once acoustic channels open at the lung surface, it becomes important to study the impact of changing alveoli diameter and spacing. In fact, these parameters influence waves capabilities of propagating beyond the pleural-line and how the different contributions add up. The tested numerical domain is of size 4 cm $\times$ 1.6 cm, the investigated frequencies vary from 1 to 5 MHz, the diameters and spacing range from 100 to 400 $\mu$m and from 20 to100 $\mu$m, respectively. Results show the strong and entangled relation between the wavelength, the domain geometries and artifact visualization, allowing to better understand propagation in such a complex medium and opening several possibilities for future studies.

### *Invited Paper*

**3:55**

**4pBAb4. Simulating lung ultrasound imaging with body wall and alveolar anatomy.** Danai Soulioti (Biomedical Eng., Univ. of North Carolina at Chapel Hill and North Carolina State Univ., Chapel Hill, NC), Oleksii Ostras (None, Chapel Hill, NC), and Gianmarco Pinton (Biomedical Eng., Univ. of North Carolina at Chapel Hill and North Carolina State Univ., 116 Manning Dr., Mary Ellen Jones Rm. 9212A, Chapel Hill, NC 27599, gia@email.unc.edu)

The majority of diagnostic ultrasound imaging is performed in regions with modest variations in the impedance (~5%) which results in relatively small reflections from soft tissue interfaces. Lung ultrasound imaging is unique because it attempts to image an air-filled organ which is almost totally reflective. Thus the interpretation of lung ultrasound imaging depends on the analysis of the scattering, multiple scattering, and reverberation physics occurring at or near the complex soft-tissue/air interfaces. Even though this physics determines the image content, its relationship to the image, and its dependence on the diseased states of the lung are poorly understood. To establish a link between body wall and lung anatomy and the resulting ultrasound image we use a custom ultrasound simulation tool (Fullwave) in conjunction with acoustical maps of human body derived from the visible human project and maps of alveolar structure derived from lung histology. The experimentally validated simulations of ultrasound propagation includes attenuation, dispersion, nonlinearity, and the multiple interactions of sound at soft tissue/air interfaces. Ultrasound images are then simulated based on the first principles of propagation and reflection. We demonstrate that key features of clinical lung ultrasound imaging such as A- and B-lines can be observed. Anatomical and diagnostically relevant parameters, such as fluid percentage, alevolar density, are pleural thickness are varied to determine their effect on the final image.

THURSDAY AFTERNOON, 10 DECEMBER 2020 — 1:05 P.M. TO 2:35 P.M.

# Session 4pCAa

## Computational Acoustics: General Topics in Computational Acoustics III

Trevor C. Wilson, Cochair
*Mechanical and Aerospace engineering, Oklahoma State University, 2714 N Running Bear St., Stillwater, OK 74075*

D. Keith Wilson, Cochair
*Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., Hanover, NH 03755*

**Chair's Introduction—1:05**

### *Contributed Papers*

**1:10**

**4pCAa1. Using constructed impulse responses to study the influence of sound speed profiles on outdoor acoustic propagation.** Michelle E. Swearingen (Construction Eng. Res. Lab., US Army ERDC, PO Box 9005, Champaign, IL 61826, michelle.e.swearingen@usace.army.mil) and Michael J. White (Construction Eng. Res. Lab., US Army ERDC, Champaign, IL)

Time-domain predictions can be accomplished by either explicitly calculating within the time domain, such as with the Finite-Difference, Time-Domain method, or by processing frequency-domain calculations into impulse responses. While the FDTD method has been used extensively in outdoor sound propagation studies, the requirement for extensive computational resources greatly limits the computational domain. The impulse response method, while well-utilized in underwater sound propagation has seen limited use in outdoor sound studies. For a given environmental scenario, the impulse response may be obtained from the inverse Fourier transform of transmission loss simulations in the frequency-domain. When it is convolved with the time variation of the source, the receiver signal may be readily predicted. One may re-use the impulse response for any new source, so long as the sampling criteria are met, and the source and receiver positions and environment match those of the modeled situation. A discussion of the implications of frequency resolution and example comparisons to measurements are presented. [This work was funded by the Assistant Secretary of the Army (Acquisition, Logistics, and Technology) [ASA(ALT)] with portions funded under 0602784/T40/24 and 0602146/AR9/01. Distribution Statement A: Approved for public release; Distribution is unlimited.]

**1:30**

**4pCAa2. Modeling infrasound propagation from Tornado producing storms.** Trevor C. Wilson (Mech. and Aerosp. Eng., Oklahoma State Univ., 2714 N Running Bear St., Stillwater, OK 74075, Trevor.wilson@okstate.edu), Real J. KC (Mech. and Aerosp. Eng., Oklahoma State Univ., Stillwater, OK), Brian R. Elbing (Mech. and Aerosp. Eng., Oklahoma State Univ., Stillwater, OK), and Matthew S. Van Den Broeke (Earth and Atmospheric Sci., Univ. of Nebraska Lincoln, Lincoln, NE)

Tornado producing storms have been shown to emit infrasound (sound below 20 Hz) before and after tornadogenesis. This infrasound can be detected over large distances due to the low atmospheric attenuation of sound signals at low frequencies. The ability for infrasound signals to travel large distances could allow for the use of infrasound microphone arrays to assist with tornado detection and improve tornado warnings. The current work will focus on investigating the effects of the local atmosphere on a propagating infrasound signal by running simulations utilizing an atmospheric modeling code known as AVO-G2S and a collection of numerical models for the propagation of infrasound known as NCPAprop. This work will report the results from these simulations which investigated tornado and hail producing storms that occurred in Oklahoma between 2017 and 2020. Particularly, the impact of acoustic and atmospheric models on emitted infrasound signals will be investigated. [This work was funded by NOAA under Grant No. NA19OAR4590340.]

**1:50**

**4pCAa3. Modelling sonic boom propagation through planetary boundary layer turbulence near the lateral extent of the carpet.** Alexander N. Carr (Mech. and Aerosp. Eng., Univ. of Florida, PO BOX 116250, 939 Sweetwater Dr., Gainesville, FL 32611, alexcarr.1721@gmail.com), Joel B. Lonzaga (Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA), and Steven A. Miller (Mech. and Aerosp. Eng., Univ. of Florida, Gainesville, FL)

Nonlinear acoustic propagation of sonic booms in the atmospheric boundary layer is considered in the context of a one-way solution to a third order wave equation. A split-step approach is utilized to compute different physical effects efficiently. Unlike previous approaches, the diffraction effects are computed exactly in the forward direction with no restriction on the angle of propagation. This results in more accurate modelling of sonic boom near the carpet edge. Heterogenous flow effects are incorporated with a wide-angle parabolic approximation. Nonlinear propagation is computed with a Burgers-Hayes method. Turbulence in the medium is constructed by the method of random Fourier modes. The turbulence spectra are constructed using an altitude dependent 3-dimensional von Karman spectrum. The turbulence field is considered frozen, as the eddy turnover time in the atmospheric boundary layer is generally much larger than the propagation time of the acoustic wave. Comparisons with benchmark predictions from the PCBoom software are conducted. The benchmark cases provide insight into the accuracy of the current prediction code for non-turbulent atmospheres. Future work is discussed regarding the prediction of shaped booms, boundary conditions, and predictions near the lateral extent of the boom carpet where previous approaches fail to capture the diffraction effects.

**2:10**

**4pCAa4. A physics-informed neural network for sound propagation in the atmospheric boundary layer.** Chris L. Pettit (Aerosp. Eng. Dept., U.S. Naval Acad., 590 Holloway Rd., MS 11-B, Annapolis, MD 21402, pettitcl@usna.edu) and D. Keith Wilson (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

We describe what we believe is the first effort to develop a physics-informed neural network (PINN) to predict sound propagation through the atmospheric boundary layer. PINN is a recent innovation in the application of deep learning to simulate physics. The motivation is to combine the strengths of data-driven models and physics models, thereby producing a regularized surrogate model using less data than a purely data-driven model. A suitable loss function is central to training a data-driven model. In a PINN, the data-driven loss function is augmented with penalty terms for deviations from the underlying physics, e.g., a governing equation or a boundary condition. Training data are obtained from Crank-Nicholson solutions of the parabolic equation with homogeneous ground impedance and Monin-Obukhov similarity theory for the effective sound speed in the moving atmosphere. Training data are random samples from an ensemble of solutions for combinations of parameters governing the impedance and the effective sound speed. PINN output is processed to produce realizations of transmission loss that look much like the Crank-Nicholson solutions. We describe the framework assembled to implement acoustics PINN, and practical concerns related to the network architecture, the size of the training set, and the design of the physics-informed loss function.

## Session 4pCAb

# Computational Acoustics: General Topics in Computational Acoustics IV

Jennifer Cooper, Chair
*Applied Physics Laboratory, Johns Hopkins University, 11100 Johns Hopkins Rd., Mailstop 8-220, Laurel, MD 20723*

**Chair's Introduction—2:50**

## *Contributed Papers*

**2:55**

**4pCAb1. Holistic, long-term soundscape monitoring in Upstate New York using convolutional long short-term memory deep neural networks.** Mallory Morgan (Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, morgam11@rpi.edu) and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

As part of a long-term bioacoustics monitoring project, audio data containing both anthrophony and biophony was collected 24/7 in a residential area of upstate NY for ten months of the 2019–2020 year. To analyze the ecological content of the data with as little manual intervention as possible, the data is automatically classified using deep learning techniques. First, the data is segmented and fed through a binary CNN long short-term memory network to separate "signal" from "silence." Next, a small subset of the dataset is manually annotated via visual inspection of log-mel spectrograms to train a multiclass CNN-LSTM—a method which reaches testing accuracies of over 90%. Algorithm performance on this manually annotated dataset is compared to performance on unabridged, "real world" audio data, and strategies to handle issues such as lack of training data, multi-label classification, and the "none of the above" class are also explored. The classification results are ultimately used to generate long-term seasonal sound maps which are cross-referenced with local weather data.

**3:15**

**4pCAb2. Clustering geospatial data for machine learning modeling of ambient soundscapes.** Mitchell C. Cutler (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84604, mitchellccutler@gmail.com), Katrina Pedersen, Mark K. Transtrum (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Shane V. Lympany, and Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC)

Outdoor ambient acoustical environments may be predicted through supervised machine learning using geospatial features as inputs. Previous work used K-Means clustering applied to the geospatial features to identify geographic regions that are geospatially distinct. The clustering results help provide physical insights regarding which features are likely to play the largest roles in the supervised learning model and which locations are impacted by different acoustic training data. However, these results may be sensitive to details of the geospatial data, such as how the data are scaled or the presence of similar redundant features. This work builds on previous results by constructing a reduced feature set by removing redundant geospatial features and by using a physically motivated scaling scheme. Clustering analysis applied to this new dataset indicates that the contiguous United States can be naturally clustered into eight human-interpretable geographic regions. Hierarchical clustering is used to further subdivide these eight clusters into more fine-grained regions. One finding of interest is that no geospatial layer in the present soundscape model uniquely identifies rivers. These results will guide further geospatial layer development and acoustical data collection for more accurate soundscape models. [Work supported by a U.S. Army SBIR.]

**3:35**

**4pCAb3. Excess attenuation at the beach: A model validation.** Andrea Vecchiotti (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., N.E., Washington, DC 20064, vecchiotti@cua.edu), Teresa J. Ryan, Faith A. Cobb, Nia Wilson (Eng., East Carolina Univ., Greenville, NC), Joseph Vignola, and Diego Turo (Mech. Eng., The Catholic Univ. of America, Washington, DC)

This work presents a comparison between experimental results and numerical modeling of atmospheric sound transmission loss across a range that includes water and a sandy shore. This work is part of a larger project developing a numerical model of long-range ($\sim$3 km) atmospheric acoustic propagation in littoral or riverine environments with a near-shore acoustic source and on-shore receivers. The narrow angle parabolic equation method used in this work accounts for wind and temperature variation with elevation along the acoustic propagation path. The beach is modelled as an equivalent fluid employing the Johnson-Champoux-Allard-Pride-Lafarge (JCAPL) model. This eight-parameters model is reduced to a one-parameter model by considering the sand as randomly packed spherical particles. The single parameter is the grain size. Measured grain size distributions of the sand and its change in water content with the distance from the water inform model development of the beach. Sound pressure levels predicted by three model variations are compared with measurements. Advantages and drawbacks of model complexity are presented. We show that short spans of saturated sand are effectively modelled with a fully reflective surface whereas longer spans of sand require varying levels of sound absorption.

## Session 4pEA

# Engineering Acoustics: General Topics in Engineering Acoustics III

Thomas E. Blanford, Cochair
*The Pennsylvania State University, State College, PA 16804*

Caleb F. Sieck, Cochair
*Code 7160, U.S. Naval Research Laboratory, 4555 Overlook Ave. SW, Washington, D.C. 20375*

Michael R. Haberman, Cochair
*Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758*

**Chair's Introduction—1:05**

### *Contributed Papers*

**1:10**

**4pEA1. Characterizing the acoustic impedance and attenuation of biocompatible elastomers: An optimal design of experiments approach.** Valerie Rennoll (Elec. and Comput. Eng., Johns Hopkins Univ., 3400 N. Charles St., Baltimore, MD 21218, vrennol1@jhu.edu), Ian M. McLane, Mounya Elhilali, and James West (Elec. and Comput. Eng., Johns Hopkins Univ., Baltimore, MD)

Polymers have a range of densities and stiffnesses that can be further tuned with the addition of dopants to control their acoustic impedance and attenuation, important properties that impact acoustic transduction in underwater, medical, and other non-airborne applications. Previous research on impedance-matched polymers is limited in scope, focusing on a specific application and varying a limited number of variables independently. Here, we perform a more generalized study to model the relationship between multiple fabrication and characterization conditions on the acoustic impedance and attenuation of biocompatible elastomers with ceramic nanoparticle dopants. Following an I-optimal design of experiments approach, fifty samples were characterized with factors of polymer type (PDMS, Ecoflex, and Polyurethane), sample thickness, dopant density, concentration, and size, and characterization frequency and temperature. Statistical analysis revealed the main variables, as well as interactions between variables, that have a significant affect on the acoustic properties of the elastomers. The resulting statistical model specifies the conditions necessary to match the polymers to any acoustic impedance in the range of 1 to 2.5 MRayls. We demonstrate how the model is used to fabricate an elastomer with an acoustic impedance matched to skin and minimum attenuation.

**1:30**

**4pEA2. Modeling the pure bending of plates with analog circuit models.** Carter J. Childs (Acoust., Penn State Univ., 201 Appl. Sci. Bldg., The Penn State Univ., University Park, PA 16802, cjc357@psu.edu) and Stephen C. Thompson (Graduate Program in Acoust., Penn State Univ., University Park, PA)

Previous work created a way to model flexural bending of Euler-Bernoulli beams as an analog circuit model [C. J. Childs and S. C. Thompson, *J. Acoust. Soc. Am.* **145**, 1895 (2019)]. Further work has been completed to expand the modeling technique from one-dimension into two-dimensions. This allows for enhanced accuracy in modeling physical structures, including a Timoshenko bar or pure bending of plates. The modeling technique will be presented to show bending of plates with comparable accuracy and computational resource advantages, as seen with the Euler-Bernoulli model.

**1:50**

**4pEA3. Piezoelectric leaf-cell sensor for simultaneous density and compressibility measurement.** Dwight Swett (Houston Res. Ctr., Aramco Americas, 17155 Park Row Dr., Houston, TX 77084, dwight.swett@aramcoamericas.com), Robert Adams, and Max Deffenbaugh (Houston Res. Ctr., Aramco Americas, Houston, TX)

In resonant piezoelectric sensors that measure fluid density and viscosity simultaneously the underlying physical mechanism is a thin shear boundary effect. A distributed traction develops over the resonator surface dependent on the product of density and viscosity. Accordingly, these sensors have insignificant dilatational interaction with the fluid and remain insensitive to fluid compressibility. Here we describe a novel sensor geometry sensing fluid compressibility as well as density and viscosity. The sensor utilizes a symmetric piezoelectric spoke structure to excite a circumferential "cell" geometry derived from a Rhodonea, or more commonly four-leaf roses, conformal mapping having a high degree of dilatational coupling to the fluid within and around the cell due to the curvilinear geometry resonance response. The coupling between the cell structure and the fluid volume creates a standing acoustic wave in the fluid and changes the resonance response of the piezoelectric cell as the density and compressibility of the fluid change. The admittance spectrum of the piezoelectric sensor is well fit by the Butterworth-Van Dyke model for a piezoelectric transducer, and the effective mass, damping and compliance of the model are modulated by the properties of the fluid. Measurements are presented on fluids of known density, viscosity, and compressibility.

**2:10**

**4pEA4. Increasing piezoelectric effect in radially polarized soft lead zirconate titanate (PZT) by pressure treating and its practical applications.** Eric K. Aikins (Elec. and Comput. Eng., Univ. of Massachusetts, 61 Stephanie Pl., New Bedford, MA 02745, aikinse@umich.edu)

Soft PZT/NAVY type II ceramic material is commonly used for underwater acoustics sensing applications due to its softer composition and higher electromechanical coupling coefficient than Navy Type I or Type III materials but can also be used for projector applications. The effective coupling coefficient (k-eff) of soft PZT hollow cylinders can be improved from 0.33 to 0.40 (20%) under axial pressure treatment and remains stable with time. The sensitivity of dielectric constant and piezoelectric modulus to compressive stress for hard and soft tangentially polarization ceramics is shown to be less than radially polarized ceramics. Effect of planar stress in spheres at moderate and deeper depths are presented. An experimental method is

introduced that utilizes an air-pressurized chamber to deliver stress loads to the piezoceramic. The electromechanical properties (including k-eff, dielectric constant, piezoelectric modulus, and compliance) are measured dynamically using an impedance analyzer and resonance/antiresonance techniques. Pressure treated cylinders with improved k-eff is shown to be suitable for application that utilize pressure equalized (free-flooded) transducer designs and for air-backed transducer designs under moderate depths. The improved k-eff improves the power factor bandwidth of the transducer is further studied using equivalent electrical circuit model including the effect of external pressure.

THURSDAY AFTERNOON, 10 DECEMBER 2020 — 1:05 P.M. TO 1:50 P.M.

## Session 4pIDa

## Interdisciplinary: Graduate Programs in Acoustics Poster Session I

Authors will be at their posters from 1:05 p.m. to 1:50 p.m.

### *Contributed Paper*

**1:05**

**4pIDa1. Graduate programs at the University of Minnesota related to acoustics.** Kelly L. Whiteford (Psych., Univ. of Minnesota, N218 Elliott Hall, 75 East River Parkway, Minneapolis, MN 55414, whit1945@umn.edu), Peggy Nelson (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), Hubert H. Lim (Otolaryngol., Head and Neck Surgery, Univ. of Minnesota, Minneapolis, MN), Mark A. Bee (Ecology, Evolution & Behavior, Univ. of Minnesota, St. Paul, MN), and Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN)

The University of Minnesota (UMN) has graduate programs that span the areas of Animal Bioacoustics, Psychological and Physiological Acoustics, and Speech Communication. Degrees are offered in Psychology (Ph.D.), Speech-Language-Hearing Sciences (M.A. in speech-language pathology, Au.D., and Ph.D. in speech-language-hearing sciences), Biomedical Engineering (M.S. and Ph.D.), Ecology, Evolution, and Behavior (Ph.D.), and Neuroscience (Ph.D.). Faculty across departments have a shared interest in understanding how the ear and brain work together to process sound and in developing new technologies and approaches for improving hearing disorders. Located on campus is the Center for Applied and Translational Sensory Science (CATSS), which provides opportunities for interdisciplinary collaborations across departments and industry to understand how sensory impairments work. Within CATSS is the Multi-Sensory Perception Lab, which houses shared equipment, including eye trackers and electroencephalography. The NSF-funded Graduate Training Program in Sensory Science offers support to graduate students through opportunities related to research, professional development, and public outreach. The Center for Magnetic Resonance Research houses several ultrahigh field magnets, while the Center for Neural Engineering and affiliated faculty labs also house multiple neuromodulation and neurorecording devices to interact with and monitor neural activity in humans and animals.

### *Invited Papers*

**4pIDa2. Penn State's graduate program in acoustics.** Victor W. Sparrow (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, vws1@psu.edu) and Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., University Park, PA)

This poster describes faculty research areas, laboratory facilities, student demographics, successful graduates, and recent enrollment and employment trends for the Graduate Program in Acoustics at Penn State—the only program in the U.S. offering the Ph.D. in Acoustics as well as M.S. and M.Eng. degrees. The Graduate Program in Acoustics is an interdisciplinary program with faculty from a variety of academic disciplines; administratively aligned with the College of Engineering and closely affiliated with the Applied Research Laboratory. Research areas include: structural acoustics, nonlinear acoustics, architectural acoustics, signal processing, aeroacoustics, biomedical ultrasound, transducers, computational acoustics, noise and vibration control, acoustic metamaterials, psychoacoustics, and underwater acoustics. Course offerings include fundamentals of acoustics and vibration, electroacoustic transducers, signal processing, acoustics in fluid media, sound and structure interaction, digital signal processing, experimental techniques, acoustic measurements and data analysis, ocean acoustics, architectural acoustics, noise control engineering, nonlinear acoustics, outdoor sound propagation, computational acoustics, biomedical ultrasound, flow induced noise, spatial sound and three-dimensional audio, and the acoustics of musical instruments. More than 750 Penn State Acoustics graduates serve widely throughout military and government labs, academic institutions, consulting firms, and industry.

**4pIDa3. An online Master's Degree in acoustics from Penn State.** Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, dar119@psu.edu) and Victor W. Sparrow (Graduate Program in Acoust., Penn State Univ., University Park, PA)

In addition to the doctoral and master's degrees offered to resident students, the Graduate Program in Acoustics at Penn State offers online access to graduate level courses leading to the M.Eng. degree in Acoustics. Lectures are broadcast live to students scattered around the world, while archived recordings allow working students to access lectures at their convenience. Students earn the M.Eng. in Acoustics degree by completing 30 credits of coursework and writing a capstone paper. Since 1987, more than 175 distance education students have completed the M.Eng. in Acoustics degree. Many other students take individual courses as non-degree students. Courses offered online include: elements of acoustics and vibration, elements of waves in fluids, electroacoustic transducers, signal processing, acoustics in fluid media, structural acoustics, digital signal processing, aerodynamic noise, acoustic measurements and data analysis, ocean acoustics, architectural acoustics, noise control engineering, nonlinear acoustics, outdoor sound propagation, computational acoustics, biomedical ultrasound, flow induced noise, spatial sound and 3-D audio, and the acoustics of musical instruments. This poster describes the distance education experience leading to the M.Eng. degree in Acoustics from Penn State and showcases student demographics, capstone paper topics, enrollment statistics and trends, and the success of our graduates.

## *Contributed Paper*

**4pIDa4. East Carolina University Engineering: A new option in graduate acoustics education.** Teresa J. Ryan (Dept. of Eng., East Carolina Univ., 248 Slay Hall, Greenville, NC 27858-4353, ryante@ecu.edu), Jeff Foeller, Barbara Muller-Borer, Brian Sylcott, Joseph Vignola, and Diego Turo (Mech. Eng., The Catholic Univ. of America, Washington, DC)

East Carolina University (ECU), located in Greenville, NC, is a large public university located in the heart of North Carolina's coastal plain. ECU's Department of Engineering was founded in 2004 and has grown to have a faculty of 31 and a relatively stable undergraduate enrollment of approximately 500. Undergraduate students earn a B.S. in Engineering with the choice of six areas of concentration: Biomedical, Bioprocess, Electrical, Environmental, Industrial and Systems, or Mechanical Engineering. Since 2013, the department has offered undergraduate research opportunities and elective courses in vibrations and acoustics. In Fall 2019, the department enrolled the inaugural class of students for a new research-focused Master of Science in Mechanical Engineering (MSME). The MSME program offers coursework and research opportunities in acoustics and vibrations. Qualified applicants are eligible to apply for paid research fellowships and assistantships. Specific recent topics of research have included (1) dynamics of and applications for arrays of resonant attachments (such as mass detection and vibration control) and (2) the study of atmospheric acoustics in the near-shore environment. Research collaborations with The Catholic University of America, ECU's Center for Sustainability, and ECU's Water Resources Center strengthen the breadth of the experience available to ECU's MSME students.

## *Invited Paper*

**4pIDa5. Graduate acoustics program at Brigham Young University.** Brian E. Anderson (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., N251 ESC, Provo, UT 84602, tbn@byu.edu), Scott D. Sommerfeldt, Kent L. Gee (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Jonathan D. Blotter, and Scott L. Thomson (Brigham Young Univ., Provo, UT)

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



## *Contributed Papers*

**4pIDa6. Highlights of the first acoustical description of the intonation of the Spanish in Bucaramanga, Colombia.** Yeimy J. Roberto (Modern Lang. and Linguist, Florida State Univ., 625 University Way, Tallahassee, FL, Office 356, Tallahassee, FL 32306, yjr19@my.fsu.edu)

This study presents the first acoustical description of the intonation of the Spanish in Bucaramanga, a northeast city in Colombia. A total of 376 utterances from 4 speakers was analyzed, including statements and interrogatives, elicited in spontaneous, semi-spontaneous and read contexts. Intonational analysis confirms that Bucaramangan Spanish is closest to Cali's variety of Colombian Spanish; however, a wide variety was found in the realization of nuclear accents, boundary tones, and prosodic phrasing. Statements are usually realized with a low nuclear accent when neutral, but a high nuclear accent appears when they are not neutral, in some cases showing upstep features (emphatic statements) and downstep features (in the tonal boundaries of requests). Yes-No questions, as well as WH echo questions, are realized with a rising boundary tone. In all the other instances the boundary tone is falling. This study contributes to the description of the intonation of Latin-American Spanish varieties and supports Hualde and Prieto's (2016) proposal that intonation should be discussed at two levels: the phonological level, and a broad phonetic level.

**4pIDa7. Graduate study at the Catholic University of America: Towards a century of acoustics.** Joseph Vignola (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., NE, Washington, DC 20640, vignola@cua.edu), Diego Turo (Mech. Eng., The Catholic Univ. of America, Washington, DC), Shane Guan (NOAA, Silver Spring, MD), and Teresa J. Ryan (Eng., East Carolina Univ., Greenville, NC)

The Catholic University of America (CUA) has a graduate program with a long history in acoustics dating back to the early 1930s. The acoustics program moved to the School of Engineering in the 1960s when there were strong needs in underwater acoustic studies to meet U.S. Naval applications. Recent interests include acoustical engineering, acoustic metamaterials, and environmental acoustic. Currently, a variety of graduate level acoustic courses are being offered in the CUA's Mechanical Engineering Department. Students can pursue a MS in Mechanical Engineering or Ph.D. with research in acoustics or vibrations. The courses in the program include a two-course sequence in fundamentals of acoustics, and more focused courses in ocean acoustics, atmospheric acoustics, acoustic metrology, marine bioacoustics, nonlinear vibration, acoustic imaging, and acoustic metamaterials. In addition, this program enables working professionals in the Washington, DC, area to complete a graduate degree while continuing fulltime employment by offering face-to-face classes on-site at a number of off-campus sites. These sites including the Naval Surface Warfare Center, Carderock, the Army's Night Vision lab, as well as Newport News Shipbuilding. The group has active research collaborations with the US Naval Research Laboratory, NSWC Carderock, East Carolina University and, Università Politecnica delle Marche, Italy.

## *Invited Paper*

**1:05**

**4pIDa8. Studying acoustics within the College of Engineering at the University of Nebraska – Lincoln.** Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 1110 S 67th St., Omaha, NE 68182-0816, lilywang@unl.edu), Erica E. Ryherd (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE), Joseph A. Turner (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, Lincoln, NE), and Jinying Zhu (Civil and Environ. Eng., Univ. of Nebraska - Lincoln, Omaha, NE)

This poster highlights three groups in the College of Engineering at the University of Nebraska – Lincoln (UNL) in which graduate students can study and conduct research in acoustics. Dr. Lily Wang and Dr. Erica Ryherd are active in architectural acoustics and noise control (http://nebraskaacousticsgroup.org ) within the Durham School of Architectural Engineering and Construction, based at UNL's Scott Campus in Omaha. Dr. Jinying Zhu in Civil and Environmental Engineering (also based on UNL's Scott Campus in Omaha) focuses in structural acoustics, using ultrasonic waves for concrete evaluation. Dr. Joseph Turner in Mechanical and Materials Engineering (based at UNL's City Campus in Lincoln) studies ultrasound propagation through complex media for quantitative characterization of materials/microstructure (http://quisp.unl.edu ). There is an active student chapter of the Acoustical Society of America on Scott Campus that regularly schedules meetings and activities. In the past year, these have included a car stereo competition, guest lectures, and visits to local acoustic performances. The graduate-level acoustics courses and lab facilities at UNL within the College of Engineering are presented, and the research interests and achievements of our faculty, graduates, and students are highlighted.

## *Contributed Paper*

**1:05**

**4pIDa9. Graduate studies in acoustics at Northwestern University.** Jennifer Cole (Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208-0854, jennifer.cole1@northwestern.edu), Ann Bradlow, and Matthew Goldrick (Linguist, Northwestern Univ., Evanston, IL)

Northwestern University has a vibrant and highly interdisciplinary community of acousticians. Of the 13 ASA technical areas, 3 have strong representation at Northwestern: Speech Communication, Psychological and Physiological Acoustics, and Musical Acoustics. Sound-related work is conducted across a wide range of departments including Linguistics (in the Weinberg College of Arts and Sciences), Communication Sciences & Disorders, and Radio/Television/Film (both in the School of Communication), Electrical Engineering & Computer Science (in the McCormick School of Engineering), Music Theory & Cognition (in the Bienen School of Music), and Otolaryngology (in the Feinberg School of Medicine). In addition, *The Knowles Hearing Center* involves researchers and labs across the university dedicated to the prevention, diagnosis and treatment of hearing disorders. Specific acoustics research topics across the university range from speech perception and production across the lifespan and across languages, dialect and socio-indexical properties of speech, sound design, machine perception of music and audio, musical communication, the impact of long-term musical experience on auditory encoding and representation, auditory perceptual learning, and the cellular, molecular, and genetic bases of hearing function. We invite you to visit our poster to learn more about the "sonic boom" at Northwestern University!

## *Invited Paper*

**4pIDa10. Graduate research opportunities in acoustics at the University of Michigan, Ann Arbor.** Jane H. Kim (Mech. Eng., Univ. of Michigan, 2167 Stone Rd., Ann Arbor, MI 48105, janehjki@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

The University of Michigan (UM) is host to a wide array of acoustics research which encompasses many of the Technical Committees of the ASA. Within the Department of Mechanical Engineering alone, work is being done to develop better remote sensing techniques for underwater environments, to better understand the mechanics of the human cochlea, and to build metamaterials allowing for new and exotic acoustic behaviors. Within the UM Medical School, faculty and graduate students are constantly advancing techniques for diagnostic and therapeutic ultrasound procedures. In the Department of Naval Architecture and Marine Engineering, computational and experimental tools are being developed to enable better ship design. And researchers in the Linguistics Department are using the

fundamental acoustic processes of speech to learn how humans effectively communicate. And while these are only a sample of the projects taking place at Michigan, new opportunities for acoustics research and collaboration open up each semester. Combined with a rich course catalogue, first-rate facilities, and great prospects for publication, these opportunities prepare UM graduate students for careers in industry and academia alike. Go Blue!

THURSDAY AFTERNOON, 10 DECEMBER 2020 1:50 P.M. TO 2:35 P.M.

## Session 4pIDb

## Interdisciplinary: Graduate Programs in Acoustics Poster Session II

Authors will be at their posters from 1:50 p.m. to 2:35 p.m.

### *Invited Papers*

**1:50**

**4pIDb1. Graduate training opportunities in the hearing sciences at the University of Louisville.** Shae D. Morgan (Otolaryngol., Univ. of Louisville, 627 S. Preston St., Ste. 220, Louisville, KY 40202, shae.morgan@louisville.edu), Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY), and Pavel Zahorik (Univ. of Louisville, Louisville, KY)

The University of Louisville currently offers two branches of training opportunities for students interested in pursuing graduate training in the hearing sciences: A Ph.D. degree in experimental psychology with concentration in hearing science, and a clinical doctorate in audiology (Au.D.). The Ph.D. degree program offers mentored research training in areas such as psychoacoustics, speech perception, spatial hearing, and multisensory perception, and guarantees students four years of funding (tuition plus stipend). The Au.D. program is a 4-year program designed to provide students with the academic and clinical background necessary to enter audiologic practice. Both programs are affiliated with the Heuser Hearing Institute, which, along with the University of Louisville, provides laboratory facilities and clinical populations for both research and training. An accelerated Au.D./Ph.D. training program that integrates key components of both programs for training of students interested in clinically based research is under development. Additional information is available at: http://louisville.edu/medicine/degrees/audiology and http://louisville.edu/psychology/graduate/vision-hearing.

**1:50**

**4pIDb2. Graduate programs in acoustics at the University of Salford, United Kingdom.** Jonathan A. Hargreaves (Acoust. Res. Group, Univ. of Salford, Newton Bldg., Salford M5 4WT, United Kingdom, j.a.hargreaves@salford.ac.uk)

Salford is one of the UK's leading centers for research and teaching in acoustics and audio. The courses are delivered by a research-active academic team at the very pinnacle of the profession, with an attractive staff to student ratio. Our taught courses are well-established and well-respected—we have offered postgraduate programs in acoustics since 1999. By studying with us, you will be joining a community of alumni working at the heart of many leading sound and acoustics-focused companies and organizations throughout the world. In addition, we offer MPhil and PhD programs, for those wishing to follow a purely research track. Our courses are suitable for numerate science graduates, who wish to make the transfer into the sound and acoustics industry. Study can be conducted full or part-time and can be delivered on-campus or via distance learning. Building on engineering fundamentals, it provides you with the specialist expert knowledge and a practical skill set, leaving you ready to develop a career as a future leader in audio technology or environment acoustics.



**1:50**

**4pIDb3. Underwater acoustics and ocean engineering at the University of Rhode Island.** Lora Van Uffelen (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., 213 Sheets Lab., Narragansett, RI 02882, loravu@uri.edu), James H. Miller, and Gopu Potty (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

Underwater acoustics is one of the primary areas of emphasis in the Ocean Engineering Department at the University of Rhode Island, the first Ocean Engineering program in the United States. The program offers Bachelors, Masters (thesis and non-thesis options) and Ph.D. degrees in Ocean Engineering. These programs are based at the Narragansett Bay campus, providing access to a living laboratory for student learning. Some key facilities of the program are an acoustics tank, a 100-foot-long wave tank, and currently the R/V Endeavor, a UNOLS oceanographic research vessel operated by the University of Rhode Island. A new Regional Class vessel is anticipated in 2022. At the graduate level, students are actively involved in research focused in areas such as acoustical oceanography,

propagation modeling, geoacoustic inversion, marine mammal acoustics, ocean acoustic instrumentation, and transducers. An overview of classroom learning and ongoing research will be provided, along with information regarding the requirements of entry into the program.

**1:50**

**4pIDb4. The spoken language research laboratories (SLRL) at the University of Oregon.** Melissa M. Baese-Berk (Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, mbaesebe@uoregon.edu), Kaori Idemaru, Vsevolod Kapatsinski, Tyler Kendall (Univ. of Oregon, Eugene, OR), Charlotte Vaughn (Linguist., Univ. of Oregon, Eugene, OR), and Melissa A. Redford (Univ. of Oregon, Eugene, OR)

The Spoken Language Research Laboratories (SLRL) at the University of Oregon houses 5 integrated laboratories that focus on speech communication. The SLRL occupies nearly 4000 square feet and includes the following state-of-the-art facilities: 10 sound-attenuated subject running rooms; 3 sound-attenuated (clinical-type) observation rooms; 2 single-wall sound booths in another sound attenuated room; 2 waiting rooms; 1 large computer lab; 2 graduate student workrooms; and 8 offices. The SLRL is further equipped with all necessary high-quality audio and audio-visual recording equipment. Research at the SLRL ranges from work on language variation and change to first and second language acquisition to the perception and production of spoken language. The labs provide a dynamic, supportive environment for collaborative research and training in experimental design, acoustic and speech movement analysis, statistical analysis, grant writing, science communication and community outreach. The SLRL is embedded in a vibrant language research community at the University of Oregon (UO) that is anchored in the Linguistics Department, but spread over multiple departments, including East Asian Languages & Literatures, Romance Languages & Literatures, and Communication Disorders & Sciences. The UO and the SLRL provide a myriad of research opportunities for students interested in spoken language behavior.

**1:50**

**4pIDb5. Graduate studies in acoustics and noise control in the School of Mechanical Engineering at Purdue University.** Patricia Davies (Ray W. Herrick Labs., Ray W. Herrick Labs., Purdue Univ., 177 S Russell St., West Lafayette, IN 47907, daviesp@purdue.edu), J. S. Bolton (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., West Lafayette, IN), Yangfan Liu, and Kai Ming Li (Ray W. Herrick Labs., Purdue Univ., West Lafayette, IN)

The acoustics community at Purdue University will be described with special emphasis on the graduate program in Mechanical Engineering. Around 30 Purdue faculty study aspects of acoustics and closely related disciplines and so there are many classes to choose from as graduate students structure their plans of study to complement their research activities and to broaden their understanding of acoustics. In Mechanical Engineering, the primary emphasis is on understanding noise generation, noise propagation, and the impact of noise on people, as well as development of noise control strategies including active noise control, experimental techniques, and noise and noise impact prediction tools. The noise control research is conducted at the Ray W. Herrick Laboratories, which houses several large acoustics chambers that are designed to facilitate testing of a wide array mechanical systems, reflecting the Laboratories' long history of industry-relevant research. Complementing the noise control research, Purdue has vibrations, dynamics and electromechanical systems research programs and is home to a collaborative group of engineering and psychology professors who study human perception and its integration into engineering design. There are also very strong ties between ME acoustics faculty and faculty in Biomedical Engineering and Speech Language and Hearing Sciences.

**1:50**

**4pIDb6. Graduate acoustics education in the Cockrell School of Engineering at The University of Texas at Austin.** Megan S. Ballard (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), Michael R. Haberman (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX), Neal A. Hall (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX), Mark F. Hamilton (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX), Marcia Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Tyrone M. Porter (Dept. of Biomedical Eng., The Univ. of Texas at Austin, U.S., TX), and Preston S. Wilson (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, 204 East Dean Keeton St., Mail Stop: C2200, Austin, TX 78712-0292, pswilson@mail.utexas.edu)

While graduate study in acoustics takes place in several colleges and schools at The University of Texas at Austin (UT Austin), including Communication, Fine Arts, Geosciences, and Natural Sciences, this poster focuses on the acoustics program in Engineering. The core of this program resides in the Departments of Mechanical Engineering (ME) and Electrical and Computer Engineering (ECE). Acoustics faculty in each department supervise graduate students in both departments. One undergraduate and nine graduate acoustics courses are taught in ME and ECE. Instructors for these courses include staff at Applied Research Laboratories at UT Austin, where many of the graduate students have research assistantships. The undergraduate course, taught every fall, begins with basic physical acoustics and proceeds to draw examples from different areas of engineering acoustics. Three of the graduate courses are taught every year: a two-course sequence on physical acoustics, and a transducers course. The remaining six graduate acoustics courses, taught in alternate years, are on nonlinear acoustics, underwater acoustics, ultrasonics, architectural acoustics, wave phenomena, and acoustic metamaterials. An acoustics seminar is held most Fridays during the long semesters, averaging over ten per semester since 1984. The ME and ECE departments both offer Ph.D. qualifying exams in acoustics.

**1:50**

**4pIDb7. Graduate research and education in architectural acoustics at Rensselaer Polytechnic Institute.** Ning Xiang (School of Architecture, Rensselaer Polytechnic Inst., 110 Eighth St., Troy, NY 12180, xiangn@rpi.edu), Jonas Braasch, and Todd Brooks (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

The Graduate Program in Architectural Acoustics has been constantly advanced from its inception in 1998 with an ambitious mission of educating future experts and leaders in architectural acoustics, due to the rapid pace of change in the fields of architectural-, physical-, and psycho-acoustics. Recent years the program's pedagogy using "STEM" (science, technology, engineering, and mathematics) methods has been proven to be effective and productive, including intensive, integrative hands-on experimental components that integrate architectural acoustics theory and practice. The graduate program has recruited graduate students from a variety of disciplines including individuals with B.S., B.Arch. or B.A. degrees in Mathematics, Physics, Engineering, Architecture, Electronic Media, Sound Recording, Music and related fields. Graduate students under this pedagogy and research environment have been succeed in the rapidly changing field. RPI's Graduate Program in Architectural Acoustics has since graduated more than 120 graduates with both M.S. and Ph.D. degrees. Under the guidance of the faculty members they have also actively contributed to the program's research in architectural acoustics, communication acoustics, psycho-acoustics, signal processing in acoustics as well as our scientific exploration at the intersection of cutting edge research and traditional architecture/music culture. This paper illuminates the evolution and growth of the graduate program.

## *Contributed Paper*

**1:50**

**4pIDb8. Masters course in environmental and architectural acoustics in London: A route on to our Ph.D. programme.** Stephen Dance (School of the Built Environment and Architecture, London South Bank Univ., Borough Rd., Torquay Rd., London SE1 0AA, United Kingdom, dances@lsbu.ac.uk)

The Masters program in Environmental and Architectural Acoustics (MSc) from London South Bank University is taught at the School of the Built Environment and Architecture. This was the first Masters course at the University and has been running for more than 40 years. The MSc program is delivered on a two and four semester basis depending if taken on a full-time or part-time basis with a two semester exemption if the applicant has already been award an Institute of Acoustic Diploma. The course is focused on the application of tools to solve real world acoustic problems in the built environment. The Masters students spend 50% of the time in the laboratory undertaken practical's which either prove or disprove classical acoustic theory using the very latest acoustics equipment. The course culminates with a thesis which the student normally undertakes over the summer. The best dissertations are then put forward for international awards and then students present their work at national or international conference

THURSDAY AFTERNOON, 10 DECEMBER 2020 — 3:35 P.M. TO 4:20 P.M.

# Session 4pIDc

# Interdisciplinary, Speech Communication, and Psychological and Physiological Acoustics: Acoustics in the COVID-19 Pandemic

Adam Maxwell, Chair
*Urology, University of Washington, Department of Urology, University of Washington, Seattle, WA 98195*

## *Contributed Papers*

**4pIDc1. Effects of face masks and speaking style on audio-visual speech perception and memory.** Sandie Keerstock (Linguist, Univ. of Texas at Austin, 305 E. 23rd St. CLA 4.400 E9, Mail Code B5100, Austin, TX 78712, keerstock@utexas.edu), Kirsten Meemann, Sarah M. Ransom, and Rajka Smiljanic (Linguist, Univ. of Texas at Austin, Austin, TX)

Though necessary, protective mask wearing in response to the Covid-19 pandemic presents communication challenges. Speech produced through these masks is quieter, muffled, and lacks important visual cues for listeners. The present study examines how such loss of visual information and signal degradation affects intelligibility and auditory memory of non-native speech. Additionally, we examined whether clear speaking style can alleviate some perceptual difficulty for masked speech. One non-native English speaker recorded short, educationally relevant video clips in three conditions: casual speech without a mask, and casual and clear speech with a mask. In an online study, native English listeners were presented with 15 video clips in each of the three mask / style conditions and transcribed what they heard. Following each condition, listeners answered questions about the content of the clips. Word recognition and memory accuracy will be analyzed. Detailed acoustic analyses will be conducted to assess how face masks and speaking style affect speech production. The results will allow us to quantify speech communication challenges arising from the widespread use of masks. The findings will have implications for communication in classrooms and hospitals where listeners have to understand teachers and healthcare providers, often time non-native speakers, through their protective barriers.

**4pIDc2. Impact of social distancing on persons with sensory loss during COVID-19 pandemic.** Peggy Nelson (Ctr. for Applied/Translational Sensory Sci., Univ. of Minnesota, Minneapolis, MN, nelso477@umn.edu), Walter Yueh-Hsun Wu, Kristi Oeding, Elizabeth Anderson, Katherine Teece, and Gordon Legge (Ctr. for Applied/Translational Sensory Sci., Univ. of Minnesota, Minneapolis, MN)

Recent mandates for social distancing due to the COVID-19 pandemic have had significant effects on the lives of older persons. Social distancing might have unequal effects on the lives of people with hearing impairment and vision impairment, possibly affecting their emotional well-being. We remotely surveyed 55 persons between the ages of 57–80 years (13 who were legally blind, 24 with significant hearing loss, and 18 healthy controls.) Initial surveys took place in April as shutdowns were mandated; these included current and retrospective responses concerning pre-pandemic conditions. Subsequent surveys of current experiences were conducted approximately monthly. Survey items included: number of weekly in-person and electronic interactions, satisfaction with access to communication, worry/concern levels, and effects of physical distancing and mask wearing on communication and stress. We noted an initial substantial reduction of in-person interactions for all groups, remaining low throughout. While remote electronic communications increased initially, electronic interactions for persons with sensory loss remained steady over time and for some these even decreased. Persons with hearing loss reported significant communication challenges due to mask wearing; persons with vision loss reported significant loss of connection due to physical distancing. Results show both widespread effects and specific effects on those with sensory loss.

**4pIDc3. Assessing how face masks affect speech intelligibility.** Melissa Randazzo (Commun. Sci. & Disord., Adelphi Univ., 158 Cambridge Ave., Garden City, NY 11530, mrandazzo@adelphi.edu), Ryan Priefer (Commun. Sci. & Disord., Adelphi Univ., Garden City, NY), and Laura Koenig (Haskins Labs, New Haven, NY)

To limit transmission of Covid-19, the Centers for Disease Control recommends covering the mouth and nose with a mask while interacting in public. Although face masks protect public health by containing respiratory aerosols and droplets, they can hinder speech perception by filtering the acoustic signal. Here, we assess how masks affect speech intelligibility in audio-only conditions. Four young adult native speakers of American English, naûve to the purpose of the experiment, produced short low-probability sentences in four conditions: dual-layer cloth, disposable surgical, N95, and no mask. Multitalker babble was applied to the stimuli. Listeners recruited through Amazon Mechanical Turk rated the intelligibility of the sentences, with speaker and mask condition counterbalanced. Along with obtaining overall intelligibility ratings as a percentage of words recognized, we will also compute intelligibility for consonants that are known to be differentiated by high-frequency spectral cues, and assess the long-term average spectra for the different mask types. We hope that the information from this study will help us understand the impact of masks on speech perception. Going forward, we will carry out parallel experiments with older individuals and those with hearing loss.

**4pIDc4. Multiclass sound event detection for respiratory disease diagnosis.** Drew Grant (Elec. and Comput. Eng., Johns Hopkins Univ., 3400 N Charles St., Baltimore, MD 21218, drewedwardgrant@gmail.com), Ian McLane, and James West (Elec. and Comput. Eng., Johns Hopkins Univ., Baltimore, MD)

Recordings from daily life contain a mix of sound types such as speech, coughing sounds, breathing sounds, and environmental noise, which provide a wealth of information related to linguistic messages, acoustic scenes, and health. These sounds in their unprocessed form overlap in both time and frequency domains, posing challenges to current methods for information extraction and analysis. To address this challenge, we introduce a novel multiclass sound event detection system that discriminates between speech, coughs, breathing, and other miscellaneous sounds (e.g., dogs barking, toilets flushing, babies crying) in the Coswara database, a crowdsourced database using a website application launched in response to COVID-19. This method extracts a feature set that includes Zero Crossing Rates, Short-Term Energy, and Mel-Frequency Cepstral Coefficients and uses Random Forests for multiclass classification. Preliminary results of the system yield a balanced accuracy of 87.5% detecting between speech, coughs, breathing, and other miscellaneous sounds. Using the multiclass sound event detection system, a time and acoustic-mediated forced-alignment technique is employed to discriminate complex sounds in real-time. We envision that the system could be used on devices with existing information extraction methods for monitoring respiratory diseases, such as pneumonia, pertussis, and COVID-19.

**4pIDc5. Measuring the complex shear modulus of seagrass with an iPhone: A COVID-19 experiment.** Gabriel R. Venegas (Ctr. for Coastal & Ocean Mapping, Univ. of New Hampshire, 10000 Burnet Rd., Austin, TX 78758, g.venegas@unh.edu), Victoria M. Congdon (Marine Sci. Inst., The Univ. of Texas at Austin, Port Aransas, TX), Nicholas A. Torres (Mech. Eng., UT Austin, Austin, TX), Megan S. Ballard, Kevin M. Lee (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), Kenneth H. Dunton (Marine Sci. Inst., The Univ. of Texas at Austin, Port Aransas, TX), and Preston S. Wilson (Walker Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

The shear modulus of seagrass is an important, yet rarely reported, parameter used to study the resilience of seagrass communities to hydrodynamic disturbances, and to understand the propagation of acoustic waves through seagrass meadows. Two intact *Thalassia testudinum* seagrass bundles were sampled from Redfish Bay (RFB) and Lower Laguna Madre (LLM) in South Texas, transported under oxygenated salt water to Austin, Texas, where measurements were performed at the primary author's residence during the COVID-19 shelter-in-place order in May 2020. A subsample of ten seagrass blades from each bundle was chosen, and the average thickness and width of each blade were recorded. Next, an iPhone was suspended from a ledge by each blade and manually subjected to an initial angular displacement with the axis of rotation colinear to the blade length. Once released, the angular velocity of the underdamped system was measured by the iPhone's gyroscope. Finally, the angular frequency and decay rate were used to infer the complex shear modulus. The complex shear moduli reported for RFB and LLM blades are $71+3i$ MPa and $145+5i$ MPa, respectively. Results suggest epiphytic growth could both increase the loss factor and decrease the shear modulus of seagrass. [Work supported by ARL:UT IR&D.]

**4pIDc6. Acoustic effects of face masks on speech signals.** Ryan M. Corey (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 456 Coordinated Sci. Lab, 1308 West Main St., Urbana, IL 61801, corey1@illinois.edu), Uriah Jones (Industrial Design, Univ. of Illinois at Urbana-Champaign, Urbana, IL), and Andrew C. Singer (Elec. and Comput. Eng., Univ. of Illinois, Urbana, IL)

Face masks muffle speech and make communication more difficult, especially for people with hearing loss. This study examines the acoustic attenuation caused by different face masks, including medical, cloth, and transparent masks, using a head-shaped loudspeaker and a live human talker. The results suggest that all masks attenuate frequencies above 1 kHz, that attenuation is greatest in front of the talker, and that there is substantial variation between mask types, especially cloth masks with different materials and weaves. Transparent masks have poor acoustic performance compared to both medical and cloth masks. Most masks have little effect on lapel microphones, suggesting that existing sound reinforcement and assistive listening systems may be effective for verbal communication with masks.

## Session 4pMU

## Musical Acoustics: General Topics in Musical Acoustics VI (Poster Session)

Authors will be at their posters from 1:05 p.m. to 1:50 p.m.

### *Contributed Papers*

**4pMU1. Directivity of the half-dome bundengan musical instrument.** Indraswari Kusumaningtyas (Dept. of Mech. and Industrial Eng., Universitas Gadjah Mada, Jl. Grafika 2 Kampus UGM, Yogyakarta 55281, Indonesia, i.kusumaningtyas@ugm.ac.id), Raymond Christianto (Dept. of Phys., Durham Univ., Durham, United Kingdom), and Gea O. Parikesit (Dept. of Nuclear Eng. and Eng. Phys., Universitas Gadjah Mada, Yogyakarta, Indonesia)

Bundengan is a traditional musical instrument from Indonesia that has a half-dome structure. It is fitted with clipped strings and long, thin bamboo bars that generate metal-like and drum-like sounds, respectively. In the past, a bundengan would be played for the personal pleasure of its player. Driven by the needs of conservation, nowadays, its use is steered towards stage performance for a larger audience. In this work, we investigate the directivity of the bundengan as one of the important aspects to consider when setting the musical instrument on stage. We measured the sound of plucked strings from four bundengans, made by traditional and contemporary craftsmen. The sound was recorded at a radius of 1 m from the centerline of the base of the instruments and at a height of 0.5 m from the floor, covering a circular angle of 360 deg. The bundengans generally produce comparable directivity patterns. Still, a number of differences are observed, possibly due to the small variations in the size of the domes and the placements of the strings inside the domes. These directivity plots provide the possibility to assess how suitable the bundengan actually is for stage performance. [Work supported by the National Geographic Society.]

**4pMU2. Atom music: Acoustical realizations of the atomic world through sonification.** Jill A. Linz (Phys., Skidmore College, 815 N Broadway, Saratoga Springs, NY 12866, jlinz@skidmore.edu)

Atom Music is an experimental music project that uses music synthesis techniques to create sonifications of atomic structure. It was first presented at the 178th meeting of ASA in San Diego, CA. Several realizations of atom music as it applies to science and to art have occurred with collaborations using both the atom tones and atom scales. Atom Music has become a unique way for people to learn about atoms and molecules. The idea of creating tones unique to each element as well as musical scales also based on each element has inspired people from many disciplines. This talk will focus on two separate collaborative endeavors. The first is an artistic collaboration that began as an interactive educational experience and resulted in six separate sound experiences. The second collaboration resulted in the creation of unique tones for each element, which can then be used as an acoustical periodic chart. The uses for this are varied. Several examples of how the acoustical periodic chart can be used will be discussed.

**4pMU3. Perceptual similarity and scaling of room reverberation features: Decay time and wet-dry ratio.** Kimberly Kawczinski (Ctr. for Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., 660 Lomita, Stanford, CA 94305, kawczins@ccrma.stanford.edu), Takako Fujioka, Elliot K. Canfield-Dafilou, and Jonathan S. Abel (Ctr. for Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., Stanford, CA)

When considering the suitability of a room impulse response (RIR) for a particular musical performance, the 60dB decay time (T60) and the ratio of reverberant to direct energy (wet-dry ratio or W2D) are important features. However, the perceptual effects of these features have not been well studied; specifically, the perceptual similarity of the reverberation may be affected differently by these two features and their combination. Here, we modified measured RIRs by adjusting the T60 and W2D by set increments; care was taken to construct modifications that were both realistic and mutually distinct. Modified RIR's were convolved with choral excerpts matching the musicological context of the spaces. The resulting sounds were paired and presented to listeners who made similarity ratings. Multidimensional scaling analysis was performed to uncover perceptual warping of distance along feature axes. Our results show perceptual scales consistent with exponential scaling for both T60 and W2D. When modifications in both T60 and W2D are considered, T60 appears to take precedence in shaping listeners' sense of source distance. Additionally, W2D has a more prominent effect in the case of shorter T60s.

**4pMU4. On the characteristics of aging voices of famous trot singers.** Ik-Soo Ann (Commun. Eng., Soongsil Univ., IT College, 369 Sangdoro, Dongjakgu, Seoul, South Korea, aisbestman@naver.com) and Myungjin Bae (Commun. Eng., Soongsil Univ., Seoul, South Korea)

In Korea, there is a trot song among traditional music that mainly focuses on delivering words by voice. Most famous trot singers sing popular songs, and their popularity has lasted for decades. However, compared to song voices in their 30s and 40s, the aging voice feels dull. The song voice of an old-fashioned trot singer is hard to feel the singing of his prime due to the aging characteristics. As we get older, the spectrum bandwidth of song voices decreases. Reducing spectrum bandwidth in vocalization simplifies voices, making it difficult for listeners to fall into the song. In this paper, the characteristics of the differences were identified by comparing and analyzing the famous voice of the famous trot singer's heyday and the current voice spectrum. The voices of famous trot singers, who have been popular for more than 30 years, have been shown to have a low frequency bandwidth of $-1\%$ a year reduced, and simplified by a decrease of approximately $-27\%$ for 30 years. In conclusion, as you get older, you need to try to recover the harmonics by varying the size of the mouth.

**4pMU5. Dashboard system to track cumulative exposure to sound levels during live music instruction.** Akansha Goel (Comput. Sci. and Eng., Biomedical Eng., Univ. of North Texas, 1155 Union Circle #310440, Denton, TX 76203-5017, akanshagoel@my.unt.edu), Eeshan Joshi, Ted Kwee-Bintoro (Texas Acad. of Mathematics and Sci., Univ. of North Texas, Denton, TX), Kamakshi Gopal (Audiol. and Speech-Lang. Pathol., College of Health and Public Service, Univ. of North Texas, Denton, TX), Kris Chesky (Instrumental Studies, College of Music, Univ. of North Texas, Denton, TX), Sara Champlin (Mayborn School of Journalism, College of Liberal Arts and Social Sci., Univ. of North Texas, Denton, TX), and Mark V. Albert (Comput. Sci. and Eng., Biomedical Eng., Univ. of North Texas, Denton, TX)

During music rehearsals, students and teachers are regularly exposed to unsafe sound levels; hearing loss is a major concern among musicians. By providing real time feedback to music instructors, we can persuade them to modify their teaching methods to achieve safe listening conditions. To

accomplish this, we created a user-friendly dashboard that displays the cumulative sound energy being collected in real-time. Since there are few standards for limiting music exposure, we used NIOSH standards for industrial noise exposure as the permissible exposure limit (PEL). Circular dials on the dashboard were programmed to display the amount of exposure relative to the PEL. To aid in identifying the type of exposure, each dial represents the amount of sound accumulation in a band of frequencies relative to typical human speech: below (15Hz–85 Hz), at (85Hz–252 Hz), and above (242Hz–24 675 Hz). The dashboard also displays an equalizer representing the instantaneous microphone input. After each session, the cumulative exposure data can be sent out in email format. Inexpensive and convenient to use, this dashboard will enable music instructors to make informed decisions on how to best modify their teaching approaches to reduce the risk of hearing loss.

**4pMU6. Tonal-timbral based bicoherence analysis of a viola da gamba suite for wolf tone characterization.** Nicholas V. Scott (Open Innovation Ctr., Riverside Res., 2640 Hibiscus Way, Beavercreek, OH 45431, nscott@riversideresearch.org), Sarah Jensen (Savor Safe Foods - A Matrix Sci. Co., Fairborn, OH), and Edward Maday (Violin and Viol Maker, Woodmere, NY)

Wolf tonal and timbral modulation of excited strings on viola da gambas are an annoying occurrence having a serious impact on the overall delivery of a musical composition. Fourier spectral analysis of this phenomenon obfuscates deep understanding through its exclusive focus on Gaussianity and disregard of frequency and quadratic phase coupling. With the desire to exhume nonlinear acoustical structure, an experimental study was conducted to investigate the efficacy of bicoherence analysis in understanding wolf tonal and timbral structure existing in two viola da gambas of similar dimension but different construction tuned to A 415 Hz. Bicoherence analysis of digitally recorded notes played on the C string and sampled at 44.1 kHz revealed significant but medium bicoherence spectral levels in variable patterns surpassing 50%. These levels appeared on unique notes for each instrument, D# and F, and were independent of amplitude providing evidence of non-Gaussian wolf tone warbling which was not always clearly audible. This was possibly due to phase coupling at very low and high frequencies lying on the perimeter of the audible music range. Results have implications for improved understanding of the relationship between viola da gamba construction and acoustical characteristics.

**4pMU7. Modeling aerosol flow for singing and brass instruments.** Spencer Thulin, Binod Bhatt, Jack Quire (Phys., Whitman College, Walla Walla, WA), and Kurt R. Hoffman (Phys., Whitman College, 345 Boyer Ave., Hall of Sci., Walla Walla, WA 99362, hoffman@whitman.edu)

At the beginning of the Covid-19 pandemic in Washington State, a choir rehearsal resulted in most of the participants contracting the virus during the rehearsal. More importantly, as people consider restarting education programs, the question of safely rehearsing choirs, bands, and symphonies needs to be addressed. Guidelines based on talking, sneezing, or coughing are insufficient to provide appropriate guidance for safe environments for collective singing or instrument performance. This paper reports our initial efforts to explore the question of aerosol movement through the air for some special cases of musical performance. We explored using open-source modeling software designed to numerically solve differential equations of motion: FEniCS and OpenFOAM. We will discuss modifying Open-source CFD Software to model the airflow and resulting aerosol spray from a singing human voice. We will also discuss our progress in applying these methods to the case of a brass instrument where the mixture is traveling through a tube of various lengths.

## Session 4pNSa

# Noise, Architectural Acoustics, and Signal Processing in Acoustics: Noise Standards–Applications, Measurement Methods, Analyses, and Data Processing Involving ANSI/ASA S12 and/or ISO Noise Standards III

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

Christopher J. Struck, Cochair
*CJS Labs, 57 States Street, San Francisco, CA 94114-1401*

**Chair's Introduction—1:05**

## *Invited Papers*

**1:10**

**4pNSa1. Applications for new high-frequency impact ratings.** John Lo Verde (Veneklasen Assoc., Santa Monica, CA), Samantha Rawlings (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, srawlings@veneklasen.com), and Wayland Dong (Veneklasen Assoc., Santa Monica, California, CA)

In January 2020, a new ASTM classification standard that defines high-frequency impact ratings was published as ASTM E3222. These ratings are named High-frequency Impact Rating, or HIR, for field testing, and High-frequency Impact Insulation Class, or HIIC, for laboratory testing. These ratings describe the high-frequency region of a proposed two-rating method of evaluating impact noise isolation [LoVerde and Dong, *J. Acoust. Soc. Am.* **141** (2017)], in which low- and high-frequency components are evaluated independently. This paper addresses practical use of HIR and HIIC, focusing on applications where exclusive use of HIR or HIIC can be beneficial.

**1:30**

**4pNSa2. Update on ASTM standards on heavy hard impact testing on fitness flooring.** Matthew Golden (Pliteq, 131 Royal Group Crecent, Woodbridge, ON L4H 1X9, Canada, mgolden@pliteq.com)

Currently there are two fitness related heavy hard impact standards under development at ASTM. One is a field-based method that uses standard objects of know geometry and mass to generate and vibration that can be measured in 1/3 octave bands. The second is a laboratory-based standard that uses a calibrated drop tower to measure the force pulse, or blocked force, due to an object impacting on the flooring surface. The Institute of Acoustics in the United Kingdom and ASHRAE are both also working on fitness noise related standards as well. This presentation will cover the current state of all of these standards, as well as information as to how to get involved in their development. The author will also share his thoughts on the importance of standards, the current state of standard development and who should get involved.

**1:50**

**4pNSa3. A suggested modification to ANSI S12.51 (ISO 3741) Annex D.** Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027, jerry@jglacoustics.com)

Annex D of ANSI S12.51 (ISO 3741) addresses the issue of qualifying reverberation rooms for determining sound power of fixed sources that emit sound at discrete frequencies. At low frequencies reverberation rooms have reduced modal density which causes a highly varying (spatially) sound field, resulting in a less accurate measure of the space-average sound pressure level in the reverberant field of the room compared to higher frequency bands. Annex D sets limits on the maximum allowable standard deviation in the 1/3-octave bands from 100 Hz to 2,500 Hz, and Annex E provides limits for 50 Hz, 63 Hz, and 80 Hz. This presentation discusses the required test procedure and presents examples of real room qualification test results and how rooms can be modified to meet this requirement. One fairly unique example will also be shown where the room fails to meet the qualification limits, when in fact it truly should. A recommended modification to the data analysis process is suggest to resolve this unique problem.

**2:10**

**4pNSa4. Comparison of room acoustics parameters measured using different techniques.** José A. Nepomuceno (Akustiks, Rua Girassol, 139 cjt 12, São Paulo 05433-000, Brazil, jan@acusticaesonica.com.br), Julio Gaspar, and Marcos Reis (Acústica & Sônica, São Paulo, Brazil)

ISO Standard 3382 is used around the world to investigate the room acoustics parameters of performing arts spaces. The measured data serves as feedback for new projects; to evaluate the results with design intent, and benchmark exercises with well-known halls. It is not always practical for acoustical consultants to carry out tests using precision techniques that can include omnidirectional sources and other apparatuses that are difficult to transport from one place to another. To investigate the results obtained with different techniques, we measured the room acoustics parameters in six venues. These venues include a multipurpose theatre seating 1500, two small black-box type venues seating 150 each, one chamber music hall seating 200 and two large halls with no seats and capacity for 700 people. It is a broad sample, including halls of different volumes and shapes. At the precision technique sessions, it was used one omnidirectional source, number of samples according to ISO 3382, and swept sine signal. At the simplified technique sessions, we use balloon bursts and handclaps. The data analysis used the same software for both techniques. This article presents the results and the pros and cons of using the simplified technique.

THURSDAY AFTERNOON, 10 DECEMBER 2020 — 1:05 P.M. TO 2:35 P.M.

## Session 4pNSb

## Noise: General Topics in Noise III

S. Hales Swift, Chair
*Sandia National Laboratories, P.O. Box 5800, MS 1082, Albuquerque, NM 87123-1082*

**Chair's Introduction—1:05**

### *Contributed Papers*

**1:10**

**4pNSb1. Lombard effect, ambient noise and willingness to spend time and money in a restaurant amongst older adults.** Rachael N. Piper (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S. 6th St., Champaign, IL 61820, rnpiper2@illinois.edu), Brianna Legner, Alyse Ruda, and PASQUALE BOTTALICO (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Restaurants have become problematic over the last few decades as they are no longer just a site to grab a meal; they are a very important aspect of the social experience. Ambient noise in venues can vary in sound pressure levels adding to the overall dining experience. A hearing impairment (or loss) already creates a deficit in the quality of communicating with other individuals and many patients with normal hearing still have difficulty in noisy environments. Thirty-six older adults (60+) will be categorized into groups, based on their audiometric information, to evaluate (1) vocal effort, (2) disturbance in communication, (3) willingness to spend time and (4) money, and (5) the noise level threshold at which intelligibility breaks down in a synthesized restaurant environment. These results will represent an important step toward the establishment of "age-friendly" communities.

**1:30**

**4pNSb2. Acoustic noise measurements during neonatal magnetic resonance scans.** Hannah Kurdila (U.S. Food and Drug Administration, 10903 New Hampshire Ave., Silver Spring, MD 20993, hannah.kurdila@fda.hhs.gov), Tayeb Zaidi, Ting Zhang, Sunder Rajan, and Subha Maruvada (U.S. Food and Drug Administration, Silver Spring, MD)

The magnetic resonance (MR) scan is a standard means to image the human body that produces significant sound pressure levels. Steps have been taken to explore and mitigate this threat for the adult patient, but now, it is becoming standard practice to use MR scanners on neonates, who are believed to be more sensitive than adults to loud sounds. We present (1) a survey of the neonatal MR practices of hospitals in the DC area, (2) details of the sound levels produced during neonatal scan protocols, and (3) an appraisal of the threat level to the neonatal patient based on our findings and the current literature surrounding neonatal hearing.

1:50

**4pNSb3. Subjective quantification of music instrument practice disturbances: A machine audition based approach.** Kevin Wu (Auralpattern, 10730 NW 66th St. Apt. 407, Doral, FL 33178, k.wu@auralpattern.com), Gang Ren, and Mitsunori Ogihara (Dept. of Comput. Sci., Inst. of Data Sci. and Computing, Univ. of Miami, Coral Gables, FL)

Music instrument practice usually causes disturbance to neighbors, especially at tight living spaces with common walls or building structural transmissions. However, the severity of disturbance is difficult to quantify because of the varieties of musical sound, music content, and many subjective factors. We present three studies on the quantification of such disturbances using machine audition tools. In the first study, musical sound at a music practice room and an adjacent room is recorded and analyzed for quantifying the disturbance caused to adjacent rooms. This study builds a quantification framework for disturbance by applying musical pattern recognition tools such as pitch tracking and onset detection to imitate humans' musical perceptions. The machine audition results are correlated with subjective rating results for calibrating the disturbance scales. The second study explores the frequency-selective transmission profiles and ambient noise patterns and their correlations to machine-rated disturbance levels and subjectively rated disturbance levels. The third study explores the effect of acoustical environment on musicians. A professional musician can adapt to the acoustical environment by making adjustments to the performance he has prepared for and these adjustments form a feedback loop that affects both the practice experience and the perceived disturbance level.

2:10

**4pNSb4. On a fast hearing test method according to sub-band listening frequency.** Seonggeon Bae (Commun. Eng., Soongsil Univ., IT-college, KangNam University, Youngin Kyunggido, South Korea, sgbae@kangnam.ac.kr) and myungjin bae (Commun. Eng., Soongsil Univ., Seoul, South Korea)

Modern man is exposed to an environment where hearing is susceptible. Hearing generally worsens in the higher frequencies as the body ages. However, in the case of noise-induced hearing loss, there is a possibility of partial hearing loss. The pure sound hearing test listens to the pure tone of each frequency for 250–8 kHz sound, which is used in everyday conversation, and grasps the response of the sound, but it takes a lot of time to measure the hearing. In this paper, we propose a new method for screening hearing in the sub-band area without being examined in the hospital. We extended the pure audio hearing measurement method by voice sub-band and examined the noise-induced hearing loss by band. The sound is divided into subbands, and the number of sounds heard is applied. The sound source heard at this time is nine pure tones with the frequency increased by Mel scale from 500 to 8000 Hz. If the number of sounds perceived in the sound source heard first is greater than the number of sounds perceived in the sound source heard later, the person with partial hearing loss is determined. In 12 trials, one of them was identified as suspected partial hearing impairment. By using this method, you can conveniently and easily determine whether you are hearing impaired to maintain a healthy life.

THURSDAY AFTERNOON, 10 DECEMBER 2020 — 2:50 P.M. TO 4:20 P.M.

# Session 4pNSc

## Noise: General Topics in Noise IV

S. Hales Swift, Chair
*Sandia National Laboratories, P.O. Box 5800, MS 1082, Albuquerque, NM 87123-1082*

**Chair's Introduction—2:50**

### *Contributed Papers*

2:55

**4pNSc1. Errors and uncertainty: Why acoustical measurements and predictions can never be precise.** Robert Putnam (Putnam Acoust. Consulting, 955 Dyson Dr., Winter Springss, FL 32708, noizmanbob@aol.com) and Richard J. Peppin (Richard J. Peppin, P.E., Rockville, MD)

Throughout the disciplines of airborne sound measurement and prediction, the matter of uncertainty continues to be given short shrift. Rare indeed is the field survey report, or the community noise modeling prediction, that openly addresses the unavoidable uncertainty inherent in all such measured data, in every airborne noise prediction, or even in the measurement methodology employed. Several times over past decades the authors have reminded the community of acoustical professionals of these serious limitations. A refresher is long overdue. Herein will be reviewed (a) why absolutely every acoustical measurement anywhere under any circumstances will have a precision no better than 1 dB, (b) the careless omission of the "Confidence Interval" being employed, whenever uncertainty is addressed at all, (c) the inevitable upward creep in sound level resulting from multiple source uncertainties, and (d) the often overwhelming influences of atmospheric and vegetation inhomogeneities. The proper methods for combining these several elements of uncertainty are given.

3:15

**4pNSc2. An acoustic-based road traffic monitoring method.** Karolina Marciniuk (Multimedia Systems Dept., Gdansk Univ. of Technol., Faculty of ETI, Gdansk, Poland) and Bozena Kostek (Audio Acoust. Lab., Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, bokostek@audioakustyka.org)

The paper shows an acoustic-based road traffic monitoring method and its capabilities of utilizing it in the road system management. First, the acoustical signal acquisition process is shown. The database development and the methodology of audio signals preprocessing are described. Next, selecting appropriate parameters used to detect the presence of the vehicle

on the road is discussed. The detected events are subjected first to the statistical evaluation related to intervals between vehicles (safe distance criterion), and then classification is carried out. Several methods for classifying vehicle types based on their sound are employed, using four classes of vehicles. Additionally, the method of assessing traffic safety is presented with regard to the vehicle tire adhesion with the road surface—by acoustic detection of the presence of water and snow on the roadway. The effectiveness of 0,968 has been obtained for the type of a vehicle, whereas 93.3% for detecting humidity on the road surface. [The project is financed by the Polish National Centre for Research and Development (NCBR) from the European Regional Development Fund under the Operational Programme Innovative Economy No. POIR.04.01.04-00-0089/16 entitled: INZNAK – "Intelligent road signs for adaptive traffic control, communicating in V2X technology."]

**3:35**

**4pNSc3. Vibro-acoustic analysis of noise inside vehicles.** Prachee Priyadarshinee (Mech. Eng., National Univ. of Singapore, #21-256B, 38 College Ave. East, Singapore 138601, Singapore, a0135595@u.nus.edu), Kian-Meng Lim, and Heow Pueh Lee (Mech. Eng., National Univ. of Singapore, Singapore, Singapore)

The noise levels in tracked vehicles subjected to rugged terrain is an important consideration due to its effects on human health, crew fatigue and system reliability. This paper aims to predict the noise in the interior of a tracked vehicle numerically, at low frequencies. These numerical predictions of interior noise were then compared with experimental measurements. Structure-borne noise has a significant contribution in the low frequency range. The generated interior noise is dependent not only on the structural vibrations, but also on the acoustic resonance of the interior cavity. Therefore, vibro-acoustic analysis was conducted using the COMSOL Multiphysics software.

**3:55**

**4pNSc4. Metamaterial solutions for noise management in hospital settings.** Letizia Chisari (School of Eng. and Informatics, Univ. of Sussex, Chichester 1 Bldg., Falmer - Brighton BN1 9QJ, United Kingdom, chisariletizia@gmail.com), Lorenzo Bonoldi (Metasonixx Ltd., Hastings, United Kingdom), and Gianluca Memoli (School of Eng. and Informatics, Univ. of Sussex, Brighton, United Kingdom)

Hospital were noisy places even before the recent pandemic, but the need to save lives have made them even noisier, with potential impacts on patients and staff. In this work, we present a feasibility study on the use of acoustic metamaterials in this context. Acoustic metasurfaces of different geometries, labyrinthine or not, have been tested using a commercial package for finite-element simulations, both separately and in composite combinations. The results are compared and benchmarked with measurements, to optimise the materials' ability to control sound transmission while allowing air to flow through. A discussion on the limitations and the perspectives of this approach completes the study.

## Session 4pNSd

# Noise, Architectural Acoustics, and Signal Processing in Acoustics: Noise Standards–Applications, Measurement Methods, Analyses, and Data Processing Involving ANSI/ASA S12 and/or ISO Noise Standards IV

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

Christopher J. Struck, Cochair
*CJS Labs, 57 States Street, San Francisco, CA 94114-1401*

**Chair's Introduction—2:50**

### *Invited Paper*

**2:55**

**4pNSd1. AASHTO specifications for noise evaluations of pavement surfaces.** Dana Lodico (Illingworth & Rodkin, Inc., 1890 Gaylord St., Denver, CO 80206, dlodico@illingworthrodkin.com)

Three specifications for the noise evaluation of pavement surfaces have been developed for the American Association of State Highway Transportation Officials (AASHTO). These include (1) the On-Board Sound Intensity (OBSI) method (AASHTO T-360-16) of directly measuring noise generated at the tire-pavement interface in isolation of other noise sources, (2) the Statistical Isolated Pass-By (SIP) method (AASHTO TP 98-13) of measuring noise from isolated vehicles in existing traffic, and (3) the Continuous-Flow Traffic Time Integrate (CTIM) method (AASHTO TP 99-18) of capturing the sound from existing traffic for all vehicles on all roadway lanes. This presentation will describe and compare the procedures of the three AASHTO measurement methods used for the noise evaluation of pavements.

### *Contributed Paper*

**3:15**

**4pNSd2. Psychoacoustic and noise annoyance of hard disk drive affects acoustic ergonomics and emotional wellness of consumers.** YiChao Ma (Seagate Singapore Design Ctr., Singapore, Singapore) and Cheng S. Chin (Newcastle Univ. in Singapore, 537 Clementi Rd. #06-01, Singapore 599493, Singapore, cheng.chin@ncl.ac.uk)

Human hearing is a complex system due to the highly nonlinear nature of human ears. The ears perform many tasks simultaneously from identifying sound direction, estimating a distance from a sound source, masking of sound, and evaluating the noise. The objective sound power measurement is unable to convince the subjective nature of sound evaluation. Psychoacoustic measures such as sharpness, roughness, loudness, and fluctuation strength begin to show importance in understanding the quality of the sound from small interior such as Hard Disk Drive (HDD), which present in the most mobile consumer laptop. Two sets of jury tests have been performed to quantify the degree of HDD noise annoyance with psychoacoustic on broader age groups. The results indicate that the psychoacoustic have a close relationship with the HDD noise annoyance and age groups. The effect of laptop HDD on the user's emotional perceptions of different sound events caused by HDD's shows some participants are feeling less calm, annoyed, and excited as compared to before the jury tests.



### *Invited Papers*

**3:35**

**4pNSd3. Declaration and verification of product noise emission levels and the new ANSI/ASA S12.61 standard.** Matthew Nobile (Hudson Valley Acoust., Poughkeepsie, Poughkeepsie, NY 12603, mattnobile@verizon.net)

The primary goal of any product noise declaration (or "labeling") initiative is to enable consumers and other purchasers to make informed buying decisions. It doesn't matter if the product is relatively low-noise and the concern is one of annoyance or if the product is very loud and the concern is one of noise hazard. The purchaser still needs to know the noise level in order to make a prudent choice. This should be easy, right? We bring the product into a test lab and measure its sound power level. Then we report that number, correct? Yes, that would be true for testing and reporting a single unit. The problem comes when we are attempting to declare (label) an entire

production series ( “batch”). The 100 000 Model XYZ leaf blowers will each have slightly different levels. Suddenly we are talking about *population parameters*, *sampling*, and (because we also have to *verify* that the declared value is correct) *statistical inference* and *probability of acceptance*. It soon gets complicated. This paper will go over some of the basic concepts needed to understand product noise declarations and verifications and will describe the newly published ANSI/ASA standard S12.61 that sets forth the requirements for making these.

**3:55**

**4pNSd4. Development of the product noise rating (PNR).** Dana Lodico (Illingworth & Rodkin, Inc., 1890 Gaylord St., Denver, CO 80206, dlodico@illingworthrodkin.com)

A simplified Product Noise Rating (PNR) method is under development to provide noise level information on consumer products to the general public. Presenting PNR information in a simple and comprehensible manner would allow the general public to factor noise levels into their decisions and make more informed purchases. PNR includes a product noise rating scale with range-of-level indicators and a visual icon for presenting the PNR value. Previous work has determined PNR values for hair dryers and tower fans. This presentation will describe development efforts to date. Feedback is requested to help in the furthering of this method.

THURSDAY AFTERNOON, 10 DECEMBER 2020 — 1:05 P.M. TO 2:35 P.M.

## Session 4pPAa

## Physical Acoustics, Engineering Acoustics, Structural Acoustics and Vibration, and Biomedical Acoustics: Acoustofluidics III

Charles Thompson, Cochair
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John M. Meacham, Cochair
*Mechanical Engineering & Materials Science, Washington University in Saint Louis, 1 Brookings Dr, Jubel Hall, Rm 203K, St. Louis, MO 63130*

Kedar Chitale, Cochair
*Flodesign Sonics, Inc., 380 Main Street, Wilbraham, MA 01095*

Max Denis, Cochair
*University of the District of Columbia, 4200 Connecticut Ave. NW, Washington, D.C. 20008*

**Chair’s Introduction—1:05**

### *Contributed Papers*

**1:10**

**4pPAa1. Acoustic radiation force on a compressible spheroid.** Thomas S. Jerome (Appl. Res. Labs., The Univ. of Texas at Austin, 2721 Hemphill Park, Apt. C, Austin, TX 78705, tsjerome@utexas.edu), Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

The acoustic radiation force on a compressible spheroid is calculated using expansions of the scattered field in terms of both spherical and spheroidal wave functions. There is no restriction on the size or acoustic impedance of the spheroid, the structure of the incident field, or the orientation of the spheroid with respect to the incident field. The solution is the same as that developed previously for the radiation force on an elastic sphere, which is expressed as a summation of terms involving products of the coefficients in spherical wave expansions of the incident and scattered fields [Ilinskii *et al.*, *J. Acoust. Soc. Am.* **144**, 568–576 (2018)]. Spheroidal wave expansions are used to satisfy the boundary conditions on the compressible spheroid, and far-field asymptotes are used to relate the spheroidal and spherical wave scattering coefficients analytically. Explicit expressions for the scattering coefficients are available for spheroids with rigid or free surfaces. Results are compared with available analytical solutions for various limiting cases. The theoretical framework may be employed for objects of arbitrary shape provided the spherical wave expansion of the scattered field is known. [T.S.J. is supported by the Applied Research Laboratories Chester M. McKinney Graduate Fellowship in Acoustics.] [a]Deceased.

### 1:30

**4pPAa2. Extraordinary trapping by Gorkov potential of ordinary and vortex limited-diffraction beams.** Xudong Fan (Univ. of MS, 145 Hill Dr., University, MS 38677, xfan1@go.olemiss.edu) and Likun Zhang (Univ. of MS, Oxford, MS)

Trapping behaviors by using standing waves or focused beams have been investigated and widely used for acoustic levitation and tweezers. It is commonly understood that dense and stiff particles are repelled from pressure anti-nodes (pressure maximum) where the Gor'kov potential has a maximum. Here, the trapping behaviors of a small particle by using Gor'kov potential of ordinary and vortex non-diffracting beams are examined. Unlike the standing waves or strongly focused beams, the ordinary non-diffracting beam with proper parameters is found to have a Gor'kov potential minimum at its pressure maximum to trap a relatively dense and stiff particle therein [Fan and Zhang, *Phys. Rev. Appl.* **11**(1), 014055 (2019)]. For a vortex beam to trap a dense and stiff particle at the pressure null, the paraxiality of the beam has to be larger than some values. These seemly to be extraordinary behaviors are interpreted by the contribution of axial velocity to the Gor'kov potential. The results are applicable to beams with weak diffraction or weak focusing. Following from the results, ways to simplify methods for particle manipulation and development of acoustic tweezers are suggested.

### 1:50

**4pPAa3. A 3-D-printed acoustofluidic device for Raman spectroscopy.** Harrisson D. Santos (Phys., Universidade Federal de Alagoas, UFAL - Universidade Federal de Alagoas s/n, Av. Lourival Melo Mota - Tabuleiro do Martins, AL, Benedito Bentos 1, Maceió, Alagoas 57084040, Brazil, harrisson2011@gmail.com), Giclûnio C. da Silva (Phys., Universidade Federal de Alagoas, Maceió, Brazil), Amanda E. da Silva (Enfermagem e Farmácia, Universidade Federal de Alagoas, Maceió, lagoas, Brazil), José H. de Andrade, Carlos Jacinto da Silva (Phys., Universidade Federal de Alagoas, Maceió, Alagoas, Brazil), Magna S. Moreira (Enfermagem e Farmácia, Universidade Federal de Alagoas, Maceió, Alagoas, Brazil), Ueslen Silva, and Glauber T. Silva (Phys., Universidade Federal de Alagoas, Maceió, Alagoas, Brazil)

The development of technologies capable of determining direct information about cell status is highly required. Confocal Raman microscopy offers a singular pathway to monitoring chemical "fingerprints" of intracellular components via radiation-complex biomolecules interactions. Nevertheless, Raman spectroscopy's biological applications are still challenging due to the small probability of spontaneous light scattering events with bio-micromolecules. As a consequence, the signal-to-noise ratio is low. Acoustofluidic devices offer an ingenious solution to enhance the Raman signal by levitating the biological sample. In this work, we present a cheap 3-D-printed acoustofluidic device for confocal Raman spectroscopy. The device comprises a cylindrical cavity with a 750 $\mu$m-height and 4 mm-diameter, which operates around 1 MHz. Raman spectra of polystyrene microparticles (10 $\mu$m-diameter), levitating at 300 $\mu$m in water, were obtained by using a 785 nm excitation laser focused through a 40x objective lens. In another set of careful experiments, the device efficiently trapped 20-$\mu$m macrophages (cell line j774.A1), of which the preliminary Raman spectra will be presented. Finally, we will discuss Raman spectroscopy's trends as an optoacoustofluidic technique, as well as its applications in biotechnology and microbiology.

### 2:10

**4pPAa4. Optimizing the spray characteristics of an ultrasonic droplet generator with continuously variable operating frequency.** Li Shan (Mech. Eng. & Mater. Sci., Washington Univ. in St. Louis, 1 Brookings Dr., St. Louis, MO 63130, shanli@wustl.edu) and John M. Meacham (Mech. Eng. & Mater. Sci., Washington Univ. in St. Louis, St. Louis, MO)

Ultrasonic spraying is utilized in a diversity of industrial processes and biomedical applications, including nanomaterial synthesis and inhalation drug delivery. Ultrasonic droplet generators transfer energy into bulk liquids using acoustic waves to break the free liquid surface into extremely fine microdroplets. We have previously reported an approach that combines ultrasonic actuation, resonant operation, and acoustic wave focusing for efficient spraying of liquids (e.g., low surface tension fuels, high viscosity inks, and suspensions of biological cells). Microfabricated devices comprised a piezoelectric transducer, sample reservoir, and an array of acoustic horn structures with microscale orifices. Orifice size dictated droplet size, and the fixed reservoir geometry restricted operation to specific device resonant frequencies. Here, we incorporate a continuously variable reservoir height that enables dynamic adjustment of operating parameters (e.g., to improve spray efficiency or to tune the droplet size in real-time). Finite element analysis is used to predict the harmonic response of the system for a range of reservoir heights from 0.5 to 3 mm (corresponding to operating frequencies from 0.5 to 2.5 MHz). Spray quality is optimized regarding uniformity and stability of active nozzles for arrays with 10, 20, and 40 $\mu$m orifices, and model results are used to explain observed behaviors.

## Session 4pPAb

# Physical Acoustics, Engineering Acoustics, Structural Acoustics and Vibration, and Biomedical Acoustics: Acoustofluidics IV

Charles Thompson, Cochair
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*Flodesign Sonics, Inc., 380 Main Street, Wilbraham, MA 01095*

Max Denis, Cochair
*University of the District of Columbia, 4200 Connecticut Ave. NW, Washington, D.C. 20008*

**Chair's Introduction—2:50**

## *Contributed Papers*

**2:55**

**4pPAb1. Trap-and-release of motile microorganisms using glass-based standing surface acoustic wave devices.** Mingyang Cui (Mech. Eng. and Mater. Sci., Washington Univ. in St. Louis, 1 Brookings Dr., St. Louis, MO 63130, cuimingyang@wustl.edu) and John M. Meacham (Mech. Eng. & Mater. Sci., Washington Univ. in Saint Louis, St. Louis, MO)

Standing surface acoustic waves (SSAWs) are an effective tool for cell manipulation and isolation in many biological and biomedical applications. *Chlamydomonas reinhardtii*, a unicellular motile microorganism, is an excellent biological model to study ciliary mechanics; however, its swimming strength is a challenge for trapping and analysis by conventional non-contact methods. SSAW devices incorporating polydimethylsiloxane (PDMS) channels are unable to hold *C. reinhardtii* due to inadequate acoustic trapping forces at power levels that do not induce excessive heating. Under operating conditions that are suitable for trapping, the sample temperature increases rapidly to exceed the thermotolerance threshold of the cells leading to loss of function or death . In this study, we show that SSAW devices with wet-etched glass channels allow for gentle trap-and-release of *C. reinhardtii* using 10 and 25 MHz ultrasound while traditional SSAW devices with PDMS channels kill the cells. In addition, we use infrared thermography to confirm that PDMS-based SSAW devices reach significantly higher temperatures than glass-based devices at comparable input powers. Glass-based SSAW devices easily generate the required trapping forces while maintaining a biocompatible thermal environment.

**3:15**

**4pPAb2. Enhancing retention and recovery of particles in a standing wave chamber by modulating frequency and amplitude.** Krishna N. Kumar (Acoust., MilliporeSigma, 380 Main St., Wilbraham, MA 01095, krishna.kumar@milliporesigma.com), Tyler Campbell, Kedar Chitale, and Bart Lipkens (Acoust., MilliporeSigma, Wilbraham, MA)

Magnetic, electric, gravitational and acoustic field forces can be used to concentrate and separate particles. Here we focus on separating cells from a feed suspension using multidimensional acoustic standing waves to trap, cluster and gravitationally settle the cells. The application is concentration of cell cultures for cell therapy manufacturing. Process efficiency depends on frequency, power and settling time. Efficiency is enhanced by frequency and amplitude modulation. The acoustic standing wave is operated at its resonance frequency. Particle concentration and temperature affect the resonator frequency and may detune the system, resulting in lower retention. Exciting the field with multiple closely spaced frequencies increased particle retention between 5%–20% by changing the modulation and central frequency of the excitation signal. At the end of the retention process, the acoustic chamber is full of clusters which are sedimented into the collector. As cell clusters settle, they start dispersing in the fluid medium which impacts recovery. Use of an amplitude modulated acoustic wave during sedimentation reduces particle loss in the supernatant. As amplitude decreases, clusters start to settle. Before cells start to disperse, amplitude is increased to re-cluster the cells. This cycle repeats multiple times and increases cell recovery by 10%–30%.

**3:35**

**4pPAb3. Acoustic radiation force and torque on spheroids inside an ideal cylindrical cavity.** José P. Leão-Neto (Industrial Eng., Fed Univ Alagoas, Penedo, Alagoas, Brazil, fis.neto@gmail.com) and Glauber T. Silva (Phys., Fed Univ of Alagoas, Maceió, Brazil)

We perform a theoretical analysis of the radiation forces and torques produced on a spheroid particle by the acoustic modes of an ideal (with a rigid top and bottom and with soft or hard lateral walls) cylindrical chamber. Exact analytical expressions are obtained for a rigid particle and assuming the fluid inviscid approximation. A particular emphasis is given to the resonator's fundamental mode for which the axial and radial trap stiffnesses are calculated. The axial trap stiffness depends inversely with the cavity's height squared. We also predict that the radiation torque induces the particle to oscillate around its minor axis. The corresponding oscillation frequency is then determined. The theory is showcased for studying the dynamics of an Au microrod (modeled as an elongated spheroid) with a 20 $\mu$m-length and 2 $\mu$m-width in a water medium. In doing so, we use typical parameters of acoustofluidic devices (cavity's dimensions and energy density). The obtained results show terminal velocities of few micrometers per second, while the oscillation frequency is in the kilohertz range. In addition, a

discussion on the dynamics of micro-/nanorobots of spheroidal shape propelled by ultrasound is outlined.

**3:55**

**4pPAb4. Spatiotemporal differential partitioning for one-dimensional fast acoustic streaming.** Jeremy Orosco (Mech. and Aerosp. Eng., Univ. of California San Diego, 13754 Mango Dr., Unit 306, Del Mar, CA 92014, jrorosco@ucsd.edu) and James Friend (Mech. and Aerosp. Eng., Univ. of California San Diego, La Jolla, CA)

Historically, acoustofluidics modeling of streaming behaviors has relied heavily and almost exclusively upon formal asymptotic expansions in a relevant small parameter (e.g., the acoustic Mach number) in order to render tractable the highly nonlinear governing equations. However, the direction of modern acoustofluidics research dictates that no such order of magnitude separation between the acoustic and streaming fields can be generally relied upon—in extremal systems, formal asymptotic methods fail to properly extract the dynamics of interest. We outline progress toward the development of a theoretical approach that affords greater generality through its direct, explicit consideration and exploitation of all spatiotemporal scale disparities by partitioning differential operations. The method is applied to a one-dimensional problem of semiinfinite extent that, by convention, is classified as "fast streaming." The compressible Navier-Stokes equations are solved in an approximate successive manner in order to obtain the acoustic and streaming fields. We show that the steady multi-temporal behavior of this result is characterized by a conservation of energy across acoustic and streaming scales.

THURSDAY AFTERNOON, 10 DECEMBER 2020 — 12:00 NOON TO 12:30 P.M.

## Session 4pPP

## Psychological and Physiological Acoustics and Speech Communication: Remote Testing for Auditory and Speech Research III

Frederick J. Gallun, Cochair
*Oregon Hearing Research Center, Oregon Health and Science University, 3181 Sam Jackson Park Road, Portland, OR 97239*

G. Christopher Stecker, Cochair
*Center for Hearing Research, Boys Town National Research Hospital, 555 N 30th St., Omaha, NE 68131*

**Panel Discussion**

THURSDAY AFTERNOON, 10 DECEMBER 2020 — 1:05 P.M. TO 1:50 P.M.

## Session 4pSCa

## Speech Communication: Phonetics of Race and Gender (Poster Session)

Authors will be at their posters from 1:05 p.m. to 1:50 p.m.

**4pSCa1. Perceptual consequences of spectral manipulations for cisgender and transgender speakers.** Brandon Merritt (Speech, Lang., and Hearing Sci., Indiana Univ., 945 Basswood Circle, Bloomington, IN 47403, bmmerrit@iu.edu) and Tessa Bent (Speech, Lang., and Hearing Sci., Indiana Univ., Bloomington, IN)

Fundamental frequency and vowel formant frequencies are reportedly the most salient acoustic-phonetic cues used to judge speaker gender. However, the influence of these cues on listeners' extraction of gender and lexical information from spontaneous speech is poorly understood. This study evaluated the perceptual impacts of modifying fundamental frequency and formant frequencies in spontaneously produced speech by cisgender and transgender individuals. Spontaneous speech samples from 12 transgender (6 transfeminine and 6 transmasculine) and 12 cisgender (6 male and 6 female) speakers were included in three tasks: a speech-in-noise intelligibility task, a two-alternative forced-choice gender identification task, and a masculinity/femininity rating task, in one of two conditions: an unmodified condition or a modified condition in which fundamental and formant frequencies were manipulated to fall within either prototypical cisgender male/female range for the transgender speakers or an intermediate, gender-ambiguous range for the cisgender speakers. Results suggest shifting fundamental frequency and formant frequencies toward prototypically male/female ranges in connected speech can effectively alter listeners' gender identification judgements, but these variables alone may not be sufficient to alter perceived vocal masculinity/femininity to within typical cisgender ranges for transgender speakers. Further, strategies used by transfeminine speakers to achieve gender-congruent voice may negatively affect speech intelligibility.

**4pSCa2. Contextual sociophonetic prediction beyond the gender binary.** Justin T. Craft, Ian Calloway (Linguist, Univ. of Michigan, Ann Arbor, MI), and Dominique A. Bouavichith (Linguist, Univ. of Michigan, 611 Tappan St., #455D, Ann Arbor, MI 48109, dombouav@umich.edu)

Previous research has demonstrated social information affects listeners' linguistic decision-making. Strand and Johnson (1996) showed that imputed gender shifts listeners' sibilant category boundaries. Further research has shown sibilant identity influences listeners' binary categorization of gender, suggesting social-linguistic bidirectionality (Bouavichith *et al.*, 2019). This study extends this body of literature by investigating how sibilant categorization changes when acoustically masculinized speech is framed within differing social contexts. Participants completed a lexical decision task, where each word consisted of a synthesized sibilant onset and a naturalistic rime. In block 1, rimes were minimally manipulated; listeners were told the speaker identified as female during the recording. In block 2, rimes were masculinized; this manipulation was contextualized in condition 1 as the speaker's gender transition and in condition 2 as digital manipulation. If sensitive to the use of phonetic variation to convey social meaning, condition 1 listeners would be more likely than condition 2 listeners to adopt a categorization strategy in block 2 consistent with hearing a male-sounding voice (i.e., more likely to categorize ambiguous sibilants as /s/). As expected, condition 1 participants were more likely to categorize a sibilant as /s/ in block 2, while condition 2 participants did not differ across blocks.

**4pSCa3. Examining phonetic trends in the speech of transgender YouTubers.** Trevor M. Ramsey (Linguist. Dept., Univ. of Georgia, 210 Herty Dr., Gilbert Hall 142, Athens, GA 30602, rovert@uga.edu)

In LGBTQ research, transgender speech is understudied compared to gay and lesbian speech. Fundamental frequency (F0) and vowel space have both shown as useful in determining speaker gender (Simpson, 2009; Gaudio, 1994). Sibilants are also used to determine gender and sexuality (Zimman, 2017; Munson, 2007). This analysis of transgender YouTubers examines F0, vowel space, and sibilants and how their speech aligns with cisgender speakers. Videos of seven transgender YouTubers were transcribed and force-aligned. Measurements were taken comparing speakers within each group and with cisgender speakers. All transgender speakers produced F0 commensurate with cisgender speakers. Vowel space was plotted, showing transwomen produce an expanded vowel space when compared to transmen. Direct comparison to cisgender speakers was not undertaken. Differences in sibilant duration between groups was not borne out in the data. Spectral mean was also measured and transwomen showed ranges similar to ciswomen. Transmen were more variable in their production of sibilants. This study shows that transgender speakers produce speech that mostly aligns with cisgender speakers for the sounds measured. Though not representative of all trans people, these speakers show gendered speech is not simply physiological and further study may aid in efforts toward greater transgender equality.

**4pSCa4. That /s/ tiene *tumbao*: Investigating acoustic correlates of femme queerness in Bilingual Miami.** Christopher M. Mendoza (English, Program in Linguist, Florida Int. Univ., 3135 SW 98 CT, Miami, FL 33165, cmend116@fiu.edu)

Sociophonetic work on the production of /s/ has shown an indexical link between feminine gender expression and higher centers of gravity and negative spectral skew, corresponding to a fronted articulation of [s+] (Hazenberg, 2012; Podesva and Van Hofwegen, 2014). In this paper, I explore how Latinx identity and multilingualism complicate our understandings of [s+] as indicative of femme-coded identities. At the same time, this work also adds to the literature on the use of [s+] in communities of drag queens (Calder, 2019) by demonstrating how it gets mediated by the bilingual Latinx body during the process of feminine drag visual transformation. I examine the production of /s/ amongst 3 Cuban-American drag queens in sociolinguistic interviews conducted in English and Spanish while participants are transforming into their drag personae. The interviews were recorded and the center of gravity and skewness were calculated for each extracted token of /s/ presented during the exchanges using an automated Praat script (Boersma and Weenink, 2019). Preliminary results on center of gravity measurements indicate that /s/ is consistently produced by Miami drag queens with a fronted articulation in English, while Spanish productions have a higher range of variability.

**4pSCa5. Implied speaker attributes in standard sentence materials: Implications for sociophonetic research.** Alayo Tripp (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, tripp158@umn.edu) and Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Studies have shown that sentence intelligibility is affected by listeners' knowledge of attributes of talkers being perceived, such as their race [e.g., Babel and Russell, *J. Acoust. Soc. Am.* **137** (2015)]. Some of these studies have used the standard IEEE/Harvard sentences [*IEEE Trans. Audio Electroacoust.* **17** (1969)] which were developed to be phonetically balanced with low contextual probability. Hence, their content alone ought to not convey anything about speaker attributes. The current study tests this by examining ratings of 124 individuals of 80 written IEEE sentences, and a comparison set of 80 sentences from the internet discussion forum Reddit. The latter served as comparisons that were more likely to convey social attributes. Individuals rated the naturalness of the sentences, and whether they conveyed the gender and race of a person who might produce them. The IEEE sentences were rated less natural than the Reddit sentences, and less likely to convey race or gender. However, on average, 25% of the raters deemed each IEEE sentence to convey speaker gender, and 18% deemed them to convey speaker race. These findings caution against using the IEEE sentences as socially neutral stimuli in studies of talker attributes and speech intelligibility. [Funded by NIH.]

**4pSCa6. Influence of raciolinguistic expectations on phoneme discrimination in Spanish-English bilinguals.** Jennifer Dibbern (Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, jenniferdibbern2023@u.northwestern.edu) and Annette D'Onofrio (Linguist, Northwestern Univ., Evanston, IL)

Work has demonstrated that social expectations about a speaker's identity can affect how their speech is perceived (e.g., D'Onofrio, 2019) and that

interactional context, which is hypothesized to include non-linguistic cues like face or voice, may influence bilingual language processing (e.g., Grosjean, 2001). We unite these bodies of work to investigate how social information may be utilized in bilingual speech perception. In the United States, researchers have documented raciolinguistic ideologies around the use of Spanish, including that it is an indicator of Latinx identity (Rosa, 2016). To test whether racialized ideologies triggered language-specific processing strategies, thirty-one early or simultaneous Spanish-English bilingual speakers from the United States completed a visually primed phoneme discrimination task, categorizing bilabial stop continua (from /ba/ to /pa/). This design followed work demonstrating that Spanish-English bilinguals shifted their /b-p/ identification boundary as a function of the language they were told they'd be hearing (Gonzales *et al*., 2019). While raciolinguistic evaluations appeared to influence bilingual speech perception, they did not work the same way for each voice, suggesting the integration of multiple cues (both acoustic and social) in perception, and that the influence of raciolinguistic ideologies on speech perception may be contingent on complex aspects of the perceived speaker.

**4pSCa7. Effects of word-frequency and social evaluation on phonetic accommodation.** Shawn C. Foster (Linguist, Northwestern Univ., 2016 North Sheridan Rd., Evanston, IL 60208, shawnfoster2023@u.northwestern.edu) and Jennifer Cole (Linguist, Northwestern Univ., Evanston, IL)

Phonetic convergence between interlocutors is posited as a mechanism for long-term phonetic change (Pardo, 2006). This hypothesis requires that more frequent exposure to a phonetic variant increases the likelihood a speaker adopts said variant in their own speech. Certain instances of historical language change have, however, been theorized to represent divergence from, rather than convergence to, the speech patterns of an incoming group (Van Herk, 2008). We present the results of an artificial language learning experiment designed to investigate how social knowledge may mitigate frequency effects on phonetic convergence. Participants were exposed to nonce words produced by two speakers who differed systematically in their productions of front vowels. Participants in the experimental condition were informed that one speaker was a "native," speaker of the language and an experienced teacher, while the other was a non-native speaker. The proportional frequency with which each speaker was heard varied across conditions. Subjects' own productions of the words were recorded after exposure. Results show an interaction between frequency and social bias. Test productions moved towards the socially upweighted, less frequent variant, but not to the same extent as one that was both upweighted and more frequent.

**4pSCa8. Vowel characteristics of Black speakers' English in Southern Louisiana.** Irina Shport (Dept. of English, Louisiana State Univ., 260 Allen Hall, LSU, Baton Rouge, LA 70803, ishport@lsu.edu)

Although variation in English has been documented in Louisiana, little sociophonetic research has been conducted with Black Louisianans. In this study, acoustic characteristics of eleven vowels were examined in three groups of Black Louisianans (residents of New Orleans, New Iberia, and relocators from New Orleans to New Iberia after Hurricane Katrina) and a group of White New Iberians, five middle-aged participants per group. 240 productions of each vowel were elicited in a word-reading task. Vowel duration and formant measurements at 20%, 50%, and 80% into the vowel were examined in mixed model regression analyses. Results showed that Black speakers did not participate in /u, o/ fronting associated with White speech. All of them exhibited the African American Vowel Shift (except for the /ɪ/ vowel): their /a/ was fronted and lowered, /æ/ was fronted and raised, and /ɛ/ was raised. Some variation in vowel production existed among Black speaker groups as well; for example, New Orleanians had in-gliding /i, u/ and less diphthongized /o/ as compared to New Iberians. The non-homogeneity of Black speech and maintenance of phonetic variants of the native community by relocators is discussed in the context of the U.S. South.

**4pSCa9. Categorization of U.S. regional dialects and race from speech.** Emerson V. Wolff (Speech and Hearing Sci., Indiana Univ., 6820 N Delaware St., Indianapolis, IN 46220, evwolff@iu.edu), Tessa Bent (Speech and Hearing Sci., Indiana Univ., Bloomington, IN), and Jennifer Lentz (Indiana Univ., Bloomington, IN)

Listeners extract a range of indexical information about speakers from the acoustic speech signal. Listeners show moderate to high accuracy with the two dimensions investigated here—race and regional dialect—with generally higher accuracy for race than regional dialect. In this project, both race and regional dialect identification from the same talkers were tested when the dimensions were varied orthogonally. In a forced-choice categorization task, listeners were presented with 144 unique audio clips of monolingual English talkers from Midland, New York City, and Southern dialect regions who were either Asian, Black, or white. Stimuli were presented to listeners in two separate, counter-balanced testing blocks (i.e., one for race and one for regional dialect). Results showed higher accuracy for race identification than regional dialect, with white Midland talkers most accurately identified by race and Southern Black talkers most accurately identified by dialect. Both tasks showed significant interactions between the two indexical dimensions. The results suggest that an irrelevant indexical dimension can impact perception of the relevant indexical dimension. Further, the composite of a listener's identity and their experiences with people of various ethnicities, racial and dialect groups can shape their perception and response biases. [Work supported by Indiana University Hutton Honors College.]

## Session 4pSCb

## Speech Communication: Accent Perception and Talker Evaluation (Poster Session)

Authors will be at their posters from 1:50 p.m. to 2:35 p.m.

### *Contributed Papers*

**4pSCb1. The differential contribution of segmental and suprasegmental aspects of a foreign English accent to negative perceptions by native and non-native speakers.** Anna Maria La Franceschina (Brooklyn College, 2900 Bedford Ave., Brooklyn, NY 11210, annamlfr@gmail.com)

Foreign accents are often stereotyped as a "broken" form of the language, triggering negative reactions in native listeners—a phenomenon attributed to a general reduction in cognitive fluency. This study aims to determine the relative contributions of segmental and suprasegmental features to the perception of foreign-accented speech. Sentences in English were recorded by native speakers of Russian, Italian, and English (the latter as a control group). The recordings were manipulated to form 3 sets: naturally produced, intonation only (i.e., filtered in Praat with a low-pass Hann filter), and segmental only (i.e., foreign pronunciation was preserved in terms of consonants and vowels, but the intonation resembled that of English). Each utterance was presented to native (n = 26) and non-native English speakers (n = 32) residing in NYC and rated on several dimensions on a 1-5 Likert scale. The results support previous findings, indicating that foreign accents cause speakers to be perceived more negatively even in strongly multicultural environments. We discuss the effects of segmental/suprasegmental information in native and non-native perception and a number of language-specific patterns. This study adds to the body of work on the mechanisms underlying listeners' reactions to foreign-accented speech.

**4pSCb2. Native listeners' perceptual assessments of native and non-native speech and their associations with various speech properties.** Jieun Lee (Dept. of Linguist, Univ. of Wisconsin-Milwaukee, Milwaukee, WI 53201, jieunlee@uwm.edu), Dong Jin Kim, and Hanyong Park (Linguist, Univ. of Wisconsin-Milwaukee, Milwaukee, WI)

Listeners can evaluate different aspects of speech, but do they make judgments based on the same speech properties for native and non-native speech? We investigated this question by examining the contribution of various speech properties to American-English speaking listeners' ratings on fluency, comprehensibility, accentedness, and pleasantness of relatively short, spontaneous English utterances produced by American-English native speakers (i.e., native speech) and Korean native speakers (i.e., accented speech). The results showed that the four perceptual dimensions in both native and accented speech were distinctive regarding the contribution of different speech measurements in ratings, albeit some overlapped. In general, the normalized Pairwise Variability Index (the metrics measuring speech rhythm) was salient in native speech evaluations, whereas repair fluency measurement, which includes the instances of repetitions, replacements, reformations, hesitations, and false starts, was in accented speech evaluations. This repair fluency also influenced pleasantness evaluations of native speech but not the evaluations of the other dimensions in native speech. In addition, speech measurements affecting pleasantness evaluations of accented speech are also involved in the evaluations of fluency and comprehensibility of accented speech. In sum, this study suggests that listeners process the same perceptual dimensions in somewhat different manners in native and accented speech.

**4pSCb3. Perceptual consequences of pauses: Credibility and fluency of native and non-native speech.** Zachary N. Houghton (Linguist, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97405, znh@uoregon.edu), Misaki Kato (Linguist, Univ. of Oregon, Eugene, OR), Melissa M. Baese-Berk (Univ. of Oregon, Eugene, OR), and Charlotte Vaughn (Linguist, Univ. of Oregon, Eugene, OR)

Silent pauses are a natural part of speech and impact perception of speech. However, it is unclear how foreign accentedness influences perception of pauses. Specifically, studies have shown mixed results regarding whether listeners process pauses in native and non-native speech similarly or differently. A possible explanation for these results is that perceptual consequences of pauses differ depending on the type of processing that listeners engage in: a focus on the content/meaning of the speech versus characteristics/form of the speech. Thus, the present study examines the effect of silent pauses on listeners' perception of native and non-native speech in two different tasks, examining perceived credibility of the information conveyed in the speech, and perceived fluency of the speech. Specifically, we ask whether characteristics of silent pauses (e.g., presence or absence of a pause; syntactic location of a pause) influence listeners' perception differently for native versus non-native speech, and whether this pattern differs for perceived credibility and fluency. The results help us better understand how disfluency and foreign accent together impact listeners' perception. Further, this provides insight for how perceptual consequences of certain acoustic properties of the speech differ depending on how listeners are asked to evaluate the speech.

**4pSCb4. Effect of hesitation sound phonetic quality on perception of language fluency.** Tillena Trebon (Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, ttrebon@uoregon.edu) and Melissa M. Baese-Berk (Univ. of Oregon, Eugene, OR)

Nonnative speech differs from native speech in various ways, including different pausing patterns. There are two types of pauses: filled and unfilled. Unfilled pauses are silent. During filled pauses, speakers make a sound. Different languages use different sounds for filled pauses; this is described as phonetic quality. English speakers use "uh" to hesitate. Spanish speakers use "eh" to hesitate. When the phonetic quality of a hesitation sound is consistent with the hesitation sound used by native speakers, the hesitation sound is "native." A hesitation sound with phonetic quality inconsistent with a native speaker hesitation sound is "non-native." Production studies show that proficiency and speech community influence whether second language speakers produce native or nonnative hesitation sounds. However, no study has investigated the perceptual consequences using nonnative versus native hesitation sounds. This study investigates the effect of hesitation sound phonetic quality on perception of language fluency by comparing fluency ratings of sentences with nonnative hesitation sounds to fluency ratings of sentences with native hesitation sounds. This research answers questions such as: Do listeners judge non-native speakers to be less fluent when speakers produce non-native hesitation sounds? Is it beneficial for L2 learners to use native hesitation sounds to achieve perceived fluency?

**4pSCb5. Bilingual talker identification with spontaneous speech in Cantonese and English: The role of language-specific knowledge.** Angelina Lloy (Linguist, Univ. of BC, 2613 West Mall, Vancouver, BC V6T 1Z2, Canada, angelinalloy@gmail.com), Khia A. Johnson, and Molly Babel (Linguist, Univ. of BC, Vancouver, BC, Canada)

Previous research has shown that implicit or explicit knowledge about a language lends itself to improved talker recognition performance. In the context of this language familiarity effect, it is apparent that bilinguals are better at generalizing voice learning across their known languages compared to monolingual listeners [Orena *et al.*, *JASA*, **146** (2019)]. Other work suggests that training in an unknown language generalizes to a known language more robustly than the reverse [Winters *et al.*, *JASA* **123** (2008)]. The current study launches from these previous studies. This project uses excerpted Cantonese and English snippets from spontaneous interview speech from Cantonese-English bilinguals in a talker identification training experiment with Cantonese-English bilingual listeners and bilingual listeners with no knowledge of Cantonese (or related languages). Listeners are assigned to either Cantonese or English training, and then all listeners are tested on both Cantonese and English utterances to assess learning for the trained language and generalization to the bilingual's second language. Results of a multilingual questionnaire quantify listeners' code-switching abilities and multilingual competence, which, given prior research, should account for some individual differences. Using spontaneous productions, as opposed to read speech, improves the ecological validity of this research and broadens its implications.

**4pSCb6. Foreign language talker identification training does not generalize to new talkers.** Jayden J. Lee (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., 610 Commonwealth Ave., Boston, MA 02215, jaydenl@bu.edu), Jessica A. Tin, and Tyler K. Perrachione (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., Boston, MA)

Listeners identify talkers less accurately in a foreign language than their native language, but it remains unclear whether this is due to lack of experience identifying foreign-language talkers or whether linguistic processing reveals additional talker-specific information in speech. Here, we investigated whether two types of training improved the ability to learn to identify foreign-language talkers. Participants completed four days of talker identification training in Mandarin, an unfamiliar foreign language. Participants were assigned to either the "same-voices" condition, in which they trained on the same five voices during days 1–3, or the "different-voices" condition, in which they learned a new set of five voices on each day. Both groups learned five new voices on day 4. Talker identification accuracy improved across days 1–3 for the same-voices condition, but not for those in the different-voices condition. However, talker identification accuracy on day 4 did not differ from the day 1 baseline for either group. These results suggest that knowledge about foreign-language talkers is limited to the training set, and that, without specific linguistic knowledge, training on foreign-language talkers does not generalize to improved ability to learn to identify new foreign-language talkers.

**4pSCb7. The relationship between word error rate and perceptual judgment.** Seongjin Park (Dept. of Linguist, Univ. of Arizona, Tucson, AZ 85721, seongjinpark@email.arizona.edu) and John Culnan (Univ. of Arizona, Tucson, AZ)

The aim of the present study is to examine the relationship between word error rate (WER) from an automatic speech recognition system and perceptual judgments (foreign-accentedness, fluency, and comprehensibility) from human raters. In a previous study, Franco *et al.* (1997) used HMM-derived scores based on posterior probabilities of phone segments, and Deville *et al.* (1999) used an HMM/ANN recognition approach to show how the results of automatic speech recognition can be used for perceptual judgments. Park and Culnan (2019) showed the possibility to assimilate human raters' perceptual judgments by using neural network models only with the speech signal, and suggested that the model worked better on accentedness judgments than fluency judgments. In this study, we will examine whether WERs of English sentences produced by three language groups (American, Korean, Chinese) are significantly different, and if there is any difference, we will analyze the correlations between WER and perceptual judgments. The perceptual data used in Park and Culnan (2019) will be used for the analysis. The preliminary results of this study will be used to find important features to build more accurate automatic proficiency judgment models.

**4pSCb8. The effect of accent salience on generalization of perceptual adaptation.** Dae-yong Lee (Dept. of Linguist, Univ. of Oregon, 161 Straub Hall, 1290 University of Oregon, Eugene, OR 97403, daeyongl@uoregon.edu) and Melissa M. Baese-Berk (Dept. of Linguist, Univ. of Oregon, Eugene, OR)

Listeners often have difficulty understanding non-native accented speech. However, they are able to quickly adapt to non-native accented speech and generalize this adaptation to novel talkers. Previous studies that examined the mechanisms underlying generalization of adaptation mostly focused on acoustic similarity. That is, prior studies have demonstrated that acoustic similarity between non-native talkers in training and testing facilitates generalization. While acoustic similarity facilitates generalization, it is possible that salience of an accent (i.e., how noticeable the non-native accent is to a listener) also plays a significant role in generalization. That is, even if non-native accented talkers are acoustically similar to each other, generalization might not occur if a talker's non-native accent is not sufficiently distinct from the native-accented speech the listeners are familiar with. Thus, this study aims to examine the roles of salience of a non-native accent in generalization of perceptual adaptation to novel talkers. In the present study, native English speakers listen and transcribe English sentences read by Korean learners of English. Intelligibility (i.e., number of words transcribed) is measured to examine whether generalization to a novel non-native accented talker occurred. The results of the study will inform our understanding of generalization of adaptation to non-native accented talkers.

**4pSCb9. Faking It: How performed versus authentic southern accents impact word recognition.** Abby Walker (Virginia Tech, 181 Turner St. NW, Blacksburg, VA 24061, ajwalker@vt.edu) and Nicole DeFoor (Virginia Tech, VA)

While listeners are not necessarily good at distinguishing performed versus real regional dialects (e.g., Heaton, 2019), there is also evidence that performed dialects can be inaccurate, limited to salient variables, and/or phonetically exaggerated (e.g., Schilling-Estes, 1998; Zetterholm, 2003; Bell & Gibson, 2011). In this study we investigate how performed speech impacts word recognition, comparing listener responses to performed Southern dialects in one auditory lexical decision study versus native Southern dialects in another. In both studies, stimuli included monosyllabic words with vowels from the PRIZE, KIT, DRESS, THOUGHT, STRUT and FACE lexical sets, which have been shown to be strong markers of Southern US English (e.g., Gunter *et al.*, 2017). 123 native speakers of US English (46 Southern) heard Southern tokens from 4 women from Southwest Virginia, and 32 (15 Southern) heard Southern tokens from 6 actresses who did not identify as Southern. Interactions between the dialect of the speaker (real, performed), the dialect of the listener (Southern, Not-Southern), and the vowel in the word will be analyzed in terms of both accuracy and responses times.

**4pSCb10. Talker differences in clear and conversational speech: Perceived emotional valence.** Elizabeth D. Young (Commun. Sci. and Disord., Univ. of Utah, 390 S 1530 E, Rm. 1218, Salt Lake City, UT 84112, liz.d.young@utah.edu) and Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT)

Clear speech, a speaking style talkers adopt when talking to individuals with hearing loss, is often described as sounding "angry" (e.g., "I try to speak clearly, but when I do, he thinks I'm mad at him"). Previous research has indicated that listeners do indeed rate clear speech as "angry" or "disgusted" more often than conversational speech. However, this same research found that some talkers can increase the clarity of their speech without sounding angry. The present study expands the previous work by testing perceived emotion for 41 talkers in order to examine talker differences in the perceived emotional valence of clear and conversational speech. Perceptual ratings from 25 young adults with normal hearing will be gathered using an online experiment interface. Stimuli consist of 14 emotionally neutral sentences in both clear and conversational speaking styles from 41

talkers. Listeners will be asked to "judge the emotion you think you heard in that sentence" using a six-alternative, forced-choice paradigm for the following emotional categories: "anger," "fear," "disgust," "sadness," "happiness," and "neutral." It is anticipated that the current study will increase our understanding of how perceived emotional valence in clear and conversational speaking styles varies among talkers.

**4pSCb11. Varying intelligibility and linguistic complexity in sentence recognition in noise: Insights from older listeners.** Dorina Strori (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, dorina.strori@northwestern.edu), Pamela E. Souza (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL), and Ann Bradlow (Linguist, Northwestern Univ., Evanston, IL)

Recognition of foreign-accented speech by older listeners with hearing loss has typically been examined with linguistically simple materials (words and short sentences), which may not capture the complexity of realistic speech experiences. In this study, older listeners with varying degrees of hearing acuity recognized simple (i.e., short, mono-clausal, canonical declarative syntax) and complex (i.e., longer, multi-clausal, non-canonical syntax, and/or passive voice) sentences in broadband noise at a signal-to-noise ratio of 5 decibels. Three talkers produced the sentences: one L1-English, one high-intelligibility and one low-intelligibility L2 English talkers (based on prior intelligibility testing with young, normal hearing listeners). As anticipated, sentence recognition accuracy dropped with decreasing talker intelligibility and increasing sentence complexity. However, the effect of linguistic complexity on recognition accuracy was modulated by talker intelligibility, as reflected in a robust interaction between the two variables. Namely, accuracy dropped with increasing linguistic complexity only for the native and high intelligibility talkers. This pattern persisted after accounting for the effects of working memory and degree of hearing acuity. These findings suggest that older listeners exhibit different listening strategies for low versus high intelligibility talkers, wherein recognition benefits attributable to linguistic simplicity only emerge for speech over a threshold of intelligibility.

THURSDAY AFTERNOON, 10 DECEMBER 2020 — 2:50 P.M. TO 3:35 P.M.

## Session 4pSCc

## Speech Communication: Audiovisual Speech (Poster Session)

Authors will be at their posters from 2:50 p.m. to 3:35 p.m.

### *Contributed Papers*

**4pSCc1. Different facial cues for different speech styles in Mandarin tone articulation.** Saurabh Garg (Lang. and Brain Lab, Dept. of Linguist, Simon Fraser Univ., 442 East, 55th Ave., Vancouver, BC V5X1N4, Canada, srbh.garg@gmail.com), Lisa Tang (Digital Emergency Medicine, Faculty of Medicine, Univ. of BC, Vancouver, BC, Canada), Ghassan Hamarneh (Medical Image Anal. Lab, School of Comput. Sci., Simon Fraser Univ., Vancouver, BC, Canada), Allard Jongman, Joan Sereno (Phonet. & Psycholinguistics Lab., Dept. of Linguist, Univ. of Kansas, Kansas City, KS), and Yue Wang (Lang. and Brain Lab, Dept. of Linguist, Simon Fraser Univ., Burnaby, BC, Canada)

Research has shown that facial articulatory cues aid speech perception. However, how such cues are employed in different speech styles remains unclear. This study examined facial articulatory features of Mandarin tones in clear versus conversational speech styles produced by 20 native Mandarin speakers. Using computer-vision and image-processing techniques, keypoints representing each speaker's head, eyebrow and lips were identified on video, and their movement trajectories during tone productions were tracked. Thirty-three features based on distance, time and kinematics (e.g., velocity) were subsequently computed to characterize the movements. Random forest and t-test analyses were then conducted to identify the significant features between the two speech styles. Results reveal that, across tones, clear relative to conversational style involves greater movement distance and velocity, reaching articulatory targets faster. Individual tone analyses further indicate that the faster target approximation in clear speech occurs for contour tones (2-4) but not the flat tone 1. The increase in distance and velocity in clear speech is reflected on more features for the most dynamic tone 3 than for the other tones. These results suggest that clear-speech modifications for tones can be exhibited through facial movements, involving hyper-articulation across tones and tone-specific adjustments aligned with individual tone trajectories.

**4pSCc2. Users' head movement obtained from avatars in social VR applications.** Chloë Farr (Dept. of Linguist, Univ. of Victoria, 3084B Albany St., Victoria, BC V9A 1R5, Canada, chloe.l.farr@gmail.com), Yadong Liu, and Bryan Gick (Dept. of Linguist, Univ. of BC, Vancouver, BC, Canada)

Virtual Reality has been endorsed as a highly beneficial tool for face and hand-head gesture research [Sidenmark and Gellersen (2019), *ACM Trans. Computer-Human Interaction* **27**(4)]. Studies have shown [i.e., Xu, X. *et al.*, *J. Biomech.* **48**(4), 721–724 (2015)] that avatars accurately simulate user movements and allow researchers to obtain users' movements remotely without in-person contact or external recordings taken by the users. This study examines head movements of avatars in three popular social VR applications (Altspace, Oculus Home, and vTime XR) using Openface 2.0 [Baltrusaitis *et al.*, in 13th IEEE International Conference on Automatic Face & Gesture Recognition (FG 2018) (2018), pp. 59–66], software that measures head movements in a video. In each of these three platforms, we recorded a user making a sequence of head movements, rotating side-to-side and up-and-down to their maximum extension. Pilot results show that avatar movements in all three apps could be tracked by OpenFace 2.0. Further analysis will include which application's avatars best represent the user's real-time movements. This study provides insight into speech behaviour research and whether user movements are analysable via an avatar, without in-person contact or lab facilities under the new COVID-19 regulations.

**4pSCc3. Visual feedback and self-monitoring in speech learning via hand movement.** Yadong Liu (Linguist, Univ. of BC, 202-2720 Acadia Rd., Vancouver, BC V6T 1R9, Canada, yadong@connect.hku.hk), Pramit Saha (ECE, Univ. of BC, Vancouver, BC, Canada), and Bryan Gick (Linguist, Univ. of BC, Vancouver, BC, Canada)

While human speech learning largely relies on perceiving sounds [Brainard and Doupe, Nat. Rev. Neurosci. **1**(1), 31–40 (2000)], being able to see articulators also contributes to learning speech sounds [Gick *et al.* (2008), *Phonology Second Language Acquisition* **36**, 315–328 (2000)]. However, seeing hands is not necessarily helpful for learning new hand movements [Emmorey *et al.*, *J. Memory Lang.*, **61**(3), 398–411 (2009)]. This study investigates whether being able to see a hand, acting as a speech articulator, facilitates the learning of speech production via hand movements. Two groups of participants under different visual feedback conditions were asked to produce different vowel sequences via a system that maps hand movements to F1 and F2 of English vowels [Liu *et al.*, *Can. Acoust.* **48**(1) (2020)]. The results suggest that visual feedback contributes to the speed of speech learning and reaching vowel targets more accurately via hand movements. This study provides insight on the importance of visual feedback in monitoring speech, and supports the view that monitoring speech articulators visually can accelerate speech learning.

**4pSCc4. Visual scanning of a talking face when evaluating segmental and prosodic information.** Xizi Deng (Linguist, Simon Fraser Univ., 830 Duthie Ave., Burnaby, BC V5A 2P8, Canada, xizi_deng@sfu.ca), Henny Yeung, and Yue Wang (Linguist, Simon Fraser Univ., Burnaby, BC, Canada)

Prior work has shown that the mouth area can yield articulatory features of speech segments and durational information (Navarra *et al.*, 2010), while pitch and speech amplitude, are cued by the eyebrows and other head movements (Hamarneh *et al.*, 2019). It has been reported that adults will look more at the mouth when evaluating speech information in a non-native language (Barenholtz *et al.*, 2016). In the present study, we ask how listeners' visual scanning of a talking face is affected by task demands that specifically target prosodic and segmental information, which has not been examined by the prior work. Twenty-five native English speakers heard two audio sentences in English (the native language) or Mandarin (the non-native language) that might differ in segmental or prosodic information, or even both, and then saw a silent video of a talking face. Their task was to judge whether the video matched either the first or second audio sentence (or whether both sentences were the same).The results show that although looking was generally weighted towards the mouth, reflecting task demands, increased looking to the mouth predicted correct responses only for Mandarin trials. This effect was more pronounced in the Prosody and Both conditions, relative to the Segment condition ($p < 0.05$). The results suggest a link between mouth-looking and the extraction of speech-relevant information at both prosodic and segmental levels, but only under high cognitive load.

**4pSCc5. Audiovisual enhancement and single-word intelligibility in children's speech.** Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu)

It is nearly axiomatic that audiovisual (AV) speech is more intelligible than audio-only (A-only) speech, particularly when the speech is presented in a challenging listening environment, such as in background noise [e.g., MacLeod and Summerfield, *Br. J. Audiol.*, **2** (1987)]. Surprisingly, there has been only limited exploration of differences between individual talkers in the intelligibility advantage of their AV speech over their A-only speech. In this work, we ask whether different age groups of speakers elicit systematically different AV benefits in speech intelligibility tasks. A recent study examined whether children and adults' speech elicits similar AV benefits for sentence intelligibility (Karisny *et al.*, *J. Acoust. Soc. Am.* **146**). In that study, children's speech elicited smaller AV benefits than adults' speech, though this comparison was complicated by suboptimal choice of different signal-to-noise ratios to equate the groups' intelligibility in A-only conditions. The current study follows up on this work by examining single-word intelligibility in the same talkers, using a single SNR for both groups. Data collection is ongoing. Results will help us understand the role of individual-speaker variation on the magnitude of AV benefit.

**4pSCc6. Low-fidelity visual cues facilitate speech processing in a cocktail party scenario.** Margaret H. Ugolini (Ball Aerosp. Technologies Corp., 2610 Seventh St., Bldg 441, Wright Patterson Air Force Base, OH 45433, margaret.ugolini.ctr@us.af.mil), Eric R. Thompson (Air Force Res. Lab., Wright-Patterson AFB, OH), Zachariah N. Ennis (Ball Aerosp. Technologies Corp., Wright-Patterson AFB, OH), and Brian D. Simpson (Air Force Res. Lab., Wright-Patterson AFB, OH)

Speech comprehension is enhanced when a talker's face is visible, and amplitude modulations in speech support intelligibility. Listeners may benefit from visual speech by extracting amplitude modulation cues, which are represented by the mouth aperture of the talker. This multimodal enhancement of speech is often desirable, but the visual presentation of a talker's face is not always feasible. The present study investigated the degree to which a "low-fidelity" amplitude-modulation cue – an LED that changed in luminance with the amplitude of the speech envelope – contributes to speech perception for a target signal (a phrase from the Coordinate Response Measure, CRM) presented with two competing CRM speech phrases. Each trial consisted of 3 simultaneous speech streams of 5 sequential CRM phrases each. One stream included a target phrase (defined by a preset call sign) and originated from a location directly in front of the listener; competing sequences were placed at +/- 10 deg relative to that location. Listeners responded with the color and number associated with the target call sign. The presence of amplitude modulation cues and target timing cues enhanced performance. Further effects of cue type by signal-to-noise ratio will be discussed, as well as applications and future work.

**4pSCc7. Multimodal emotion perception: Influences of autism spectrum disorder and autism-like traits.** Megan R. Hancock (Speech and Hearing Sci., Indiana Univ. Bloomington, 1968 N Jordan Ave., Bloomington, IN 47406-1164, meghanco@iu.edu) and Tessa Bent (Speech and Hearing Sci., Indiana Univ., Bloomington, IN)

Individuals with autism spectrum disorder may exhibit impaired production and perception of facial and vocal affect cues. For example, both neurotypical and autistic adults tend to interpret affect less accurately from autistic faces and voices, with autistic listeners displaying overall lower identification accuracy. However, earlier studies compared neurotypical and autistic emotion perception under exclusively unimodal conditions (facial or verbal cues alone) or exclusively multimodal conditions in which utterances and static faces were presented sequentially. This study assesses neurotypical adults' accuracy and response time in a forced-choice emotion identification task with video-only, audio-only, and audiovisual recordings of neurotypical and autistic talkers. The study also investigates the relation between listeners' task performance and Autism-Spectrum Quotient (AQ) scores (degree of autism-like traits). Preliminary results show listeners are less accurate when identifying autistic emotional expressions, compared to neurotypical expressions, while talker type has no main effect on response time. Results also suggest a relation between AQ score and emotion-perception ability (accuracy and response time) in neurotypical respondents. This research will provide insight into emotion perception and production differences, along with implications for interpersonal functioning, in autism and under the broader autism phenotype within the neurotypical population. [Work supported by the Indiana University Hutton Honors College.]

## Session 4pSPa

# Signal Processing in Acoustics, Underwater Acoustics, Computational Acoustics, Acoustical Oceanography, and Animal Bioacoustics: Machine Learning in Acoustics V

Erin M. Fischell, Cochair
*Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, 266 Woods Hole Rd., MS 11, Woods Hole, MA 02543*

Daniel Plotnick, Cochair
*Penn State University, State College, PA 16804*

Wu-Jung Lee, Cochair
*Applied Physics Laboratory, University of Washington, 1013 NE 40th St., Seattle, WA 98105*

**Chair's Introduction—1:05**

## *Contributed Papers*

**1:10**

**4pSPa1. Integration of soft-robotics and deep learning to assess the coordinated emission and reception dynamics in hipposiderid bats.** Shuxin Zhang (Shandong University-Virginia Tech Int. Lab., School of Phys., Shandong Univ., Shanda South Rd., No. 27, Jinan, Shandong 250100, China, shuxinsduvt@yahoo.com), Xiaoyan Yin, Ruihao Wang, and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Some echolocating bat species with sophisticated biosonar systems such as the Old World leaf-nosed bats (Hipposideridae) change the shapes of their ultrasound emission (noseleaves) and reception baffles (pinnae) during the diffraction of the incoming and outgoing acoustic waves. Our prior research has shown that these noseleaf and pinna deformations are coordinated, but a functional significance for this coordination has yet to be demonstrated. Here, we combine a soft-robotic reproduction of the dynamic biosonar periphery in hipposiderid bats with deep neural networks to assess the impact of the coordination between the emission and reception dynamics. The biomimetic noseleaves and pinna are equipped with soft pneumatic actuators to accomplish life-like deformations. Control over this system allows us to reproduce not only the coordination patterns seen in bats but also other possible patterns that appear to be absent in bats. Comparing the suitability of these different coordination schemes between emission and reception could be used to assess which coordination pattern is best suited for a given task. As a first task, a direction finding paradigm has been selected for this assessment and work on identifying a network architecture that can take advantage of the dynamic nature of the echoes is currently in progress.

**1:30**

**4pSPa2. Visualization, detection and classification of Risso's and Pacific white-sided dolphins using an empirical mode decomposition-based process.** Mahdi H. Al-Badrawi (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, mhv35@wildcats.unh.edu), Yue Liang, Nicholas J. Kirsch (Elec. and Comput. Eng., Univ. of New Hampshire, Durham, NH), and Kerri D. Seger (Appl. Ocean Sci., Fairfax Station, VA)

Risso's and Pacific white-sided dolphins occupy similar habitats and have been acoustically documented in northward habitat expansion above the Aleutian archipelago. Therefore, being able to differentiate between the two, either more easily during manual analysis or through automation, is important to track their distribution and to estimate their population densities. In our previous work, a semi-blind detection and classification (SDC) process based on Empirical Mode Decomposition (EMD) showed promise for detecting and clustering vocalizations and mooring noises from several sources in the Bering Sea. In this work, we incorporated other decomposition methods into this process, such as Variational Mode Decomposition (VMD), to visualize peak and notch patterns in pulsed signals. The proposed process is being tested on four datasets that vary by location, water depth, and recording instrumentation to quantify its robustness and identify and overcome its limitations. A case study to determine the usability of EMD/VMD spectrograms in marine mammal click and buzz classification was performed by comparing peak and notch patterns of these two dolphin species. We will present the pros and cons of incorporating VMD spectrograms into the previously published EMD SDC process.

**1:50**

**4pSPa3. Machining process monitoring with vibration signal based manifold learning.** Jing Wang (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China, wangjing175@mails.ucas.ac.cn), Mingxin Hui, Bin Liu, Xun Wang, Xiaobin Cheng, and Jun Yang (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

Machining process monitoring based on vibration sensing is a growing demand in smart manufacturing. However, in real factories, massive process conditions which include thousands of shapes of workpieces and thousands of combinations of cutting parameters, such as spindle speed, feed rate and cutting depth, are designed and used in manufacturing. Manifold learning is able to extract essential and distinct features from the vibration signal and helps to monitor and recognize different process conditions. In this paper, the dataset, including slight and huge variation of cutting parameters and workpiece shapes, are collected for analysis. Different manifold learning algorithms are utilized and compared to mine the essential features and reduce the interference of non-sensitive features. The generalization ability of different manifold learning algorithms are discussed to fit the various process conditions. Convolutional neural networks are employed to evaluate the monitoring accuracy. The experimental results show that the features obtained by the manifold learning distinguish vibration signals of different cutting parameters in low dimensional space and give a protentional way to construct effective monitoring systems. The generalization ability to different workpieces and cutting parameters and its limitation are discussed.

2:10

**4pSPa4. Deep learning for acoustic signal processing for industrial noise.** Prachee Priyadarshinee (Mech. Eng., National Univ. of Singapore, #21-256B, 38 College Ave. East, Singapore 138601, Singapore, a0135595@u.nus.edu)

The aim of this paper is to provide an overview of the existing practices used for acoustic signal processing of noise of machines in various industries. There has been a surge in deep learning based methods for acoustic detection and classification of machinery fault diagnosis. This paper reviews the deep learning models, including the convolutional neural networks, the recurrent neural networks, the spiking neural networks, among the other variants of neural network models specific to industrial noise. Important applications such as sound detection, localization, directivity, source separation are discussed, to aid condition monitoring.

THURSDAY AFTERNOON, 10 DECEMBER 2020 2:50 P.M. TO 4:20 P.M.

## Session 4pSPb

## Signal Processing in Acoustics, Underwater Acoustics, Computational Acoustics, Acoustical Oceanography, and Animal Bioacoustics: Machine Learning in Acoustics VI

Erin M. Fischell, Cochair
*Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, 266 Woods Hole Rd., MS 11, Woods Hole, MA 02543*

Daniel Plotnick, Cochair
*Penn State University, State College, PA 16804*

Wu-Jung Lee, Cochair
*Applied Physics Laboratory, University of Washington, 1013 NE 40th St., Seattle, WA 98105*

**Chair's Introduction—2:50**

***Invited Paper***

**2:55**

**4pSPb1. Linking domain knowledge and machine interpretation using braid theory and cognitive sampling: Applications to underwater acoustics, and machine interpretation of melodic structures in North Indian classical music.** Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, 103 S Capitol St., Iowa City, IA 52242, ananya-sengupta@uiowa.edu), Bernice Kubicek (Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), Kevin Sobieski (Dept. of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), and Tejendra Majumdar (None, Kolkata, India)

Learning domain-interpretable features unbiased by model uncertainties is challenging due to training bias, incomplete model knowledge and human subjectivity. It is challenging to solve this dilemma due to limitations of available training data, incomplete or conflicting models, as well as lack of integrational framework that resolves model inconsistencies to build an unbiased feature dictionary. We introduce a novel sampling strategy called cognitive sampling that allows combining domain knowledge and machine interpretation to influence spectral sampling using geometric constructs such as braid, knots and links within the dataset, and thus enhance the accuracy of machine classification beyond so-called "black box" techniques. Beyond representative results from traditional applications such as underwater acoustics, we will also explore machine-driven music cognition based on melodic structures (Raags) from North Indian classical music. The Raag tradition exhibits intricate yet improvised melodic structure that cannot be easily quantified in Western staff notation due to its fluid forms and presence of microtones. Recent efforts to incorporate learning networks for machine-driven melodic interpretation have been met with partial success, the primary limitation being dearth of expert-interpretable feature engineering. Presented results will be based on underwater acoustic datasets and sarod recordings performed by Pt. Tejendra Narayan Majumdar.

**3:15**

**4pSPb2. Development of foliage echo simulations based on generative adversarial networks.** Michael Goldsworthy (Comput. Sci., Virginia Tech, 1075 Life Sci. Cir, ICTAS II (Mail Code 0917), Blacksburg, VA 24061, michaeljg@vt.edu) and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Through the use of biosonar, bats exhibit extraordinary navigation and localization capabilities, vastly outperforming any engineered sonar systems, especially while traversing complex cluttered environments, such as dense foliage. Therefore, computer simulation of foliage echoes has great potential to aid in studying and understanding bat biosonar, and thus may aid the development of technologies that deal with cluttered acoustic signals in air as well as in water. Generative Adversarial Networks (GANs) are a machine learning tool that have produced useful results in many computer vision tasks, but have only relatively recently been applied to acoustic signals. The work presented here uses GANs trained on a large dataset of recorded leaf echoes to create plausible acoustic models of tree echoes from the generation of impulse responses from single leaves placed in virtual 3-D space by tree generation algorithms. The generated foliage echoes show significant similarity to real foliage echoes and outperform previous simulation methods in terms of the realism of the results. These simulated echoes may be used to devise navigation methods that could be transferred to real robotic systems. In future work the focus will be on generating entire environment impulse responses in a variety of bat habitats.

**3:35**

**4pSPb3. Investigating the perceptual accuracy of machine-learning generated personalized head-eelated impulse responses.** Ming Yang Lee (Mech. Eng., The Cooper Union for the Advancement of Sci. and Art, 40 Cooper Square, New York, NY 10003, lee11@cooper.edu), Martin S. Lawless, and Melody Baglione (Mech. Eng., The Cooper Union for the Advancement of Sci. and Art, New York, NY)

An accurate pair of head-related impulse responses (HRIR) is necessary to render a realistic and immersive virtual auditory space. Since personalized HRIRs are challenging to measure, a method of using two machine-learning models to separately estimate the shape and time delays of the HRIRs based on an individual's anthropometric measurements of the head and pinnae is proposed. The investigation evaluates the perceptual accuracy of the resulting HRIRs with two subjective listening studies. First, an ABX test was conducted to determine if listeners could distinguish between auralizations made with an actual HRIR from the CIPIC database and the personalized HRIR based on the corresponding anthropometric measurements. In the second study, subjects performed localization tests in virtual reality to determine the perceptual accuracy of their personalized HRIR compared to an average HRIR. Each set of tests included one of three auralizations (noise, music, or speech) played in five source locations. Perceptual accuracy was evaluated by comparing the localization errors for the average and estimated HRIRs. The stimuli were repeated four times to investigate if subjects improved over time. The subjects were also asked to rate their listening experience to assess factors such as externalization and difficulty of localization.

**3:55**

**4pSPb4. Investigating speaker authentication system vulnerability to the limited duration of speech excerpts and voice cloning.** Szymon Zaporowski (Multimedia Systems, Gdansk Univ. of Technol., Narutowicza 11/12, Gdańsk 80-233, Poland, smck@multimed.org) and Andrzej Czyzewski (Multimedia Systems, Gdansk Univ. of Technol., Gdansk, Poland)

The impact of the length of the reference sample and the authentication sample to the accuracy of the speaker authentication employing deep learning architecture is tested in bank branches and discussed in the paper. The presented work focuses on testing different approaches to parameterizing voice credentials employing: MFCC, LPC, and GFCC as extracted features. Also, a mixed approach with the use of supervector containing the most important coefficients for each parameterization method was examined. For the purpose of this work, standard corpora for the authentication of speakers like VoxCeleb2 and Librispeech were used along with our own recordings. Another subject of the work was to investigate the immunity of the speaker verification system based on machine learning to attack attempts using the method of voice cloning. The impact of the duration of speech excerpts on the vulnerability to this type of attack was examined. The influence of quality and length of the generated recordings used for the attack was studied. It turned out that the results obtained depend on the acoustic conditions in bank branches, where there is quite a lot of noise coming from the work of banknote counters, the clatter of stamping documents, and conversations. [Project No. POIR.01.01.01-0092/19 entitled: "BIOPUAP—a biometric cloud authentication system" is currently financed by the Polish National Centre for Research and Development (NCBR) from the European Regional Development Fund.]

# Session 4pUW

## Underwater Acoustics: Underwater Acoustic Signal Processing

Martin Siderius, Chair
*Portland State Univ., 1600 SW 4th Avenue, Suite 260, Portland, OR 97201*

**Chair's Introduction—2:50**

### *Contributed Papers*

**2:55**

**4pUW1. Acoustic communication through well-head for an untethered autonomous well logging tool.** Huseyin R. Seren (Aramco Res. Center–Houston, Aramco Services Co., 17155 Park Row, Houston, TX 77084, huseyin.seren@aramcoamericas.com), Erjola Buzi (Aramco Res. Center–Houston, Aramco Services Co., Houston, TX), Renad Bougis (Virginia Tech, Blacksburg, VA), and Max Deffenbaugh (Aramco Res. Center–Houston, Aramco Services Co., Houston, TX)

Untethered autonomous well logging tools have recently been introduced to oilfields. Once deployed into a well, the tools autonomously travel in the well to collect data and return to surface after the mission is completed. A service operator needs to travel to the well-site and open the well cap to check if the tool has returned. This increases the operational time and cost. This can be addressed by implementing a return detector and notifying the operator. Steel well-head construction leaves acoustics as the most viable option for such detection. We demonstrated a proof of concept of acoustic based simplex communication system through well-head. A pair of identical transducers were used as transmitter and receiver that operates at 155 kHz. Transmitter piezo was positioned at the center of a water filled steel wellhead with 7-inch diameter and 0.25-in. wall thickness, while the receiver was taped outside the wellhead with grease to increase acoustic coupling. Transmission loss was measured as -57 dB which was partially compensated by using a receiver amplifier stage with a gain of 1000. Next, the transmitter will be incorporated on a miniaturized untethered well logging tool.

**3:15**

**4pUW2. Research on spatiotemporal fluctuation characteristics of underwater acoustic communication signals.** Zhichao Lv (Ocean Univ. of China, Qingdao City, China), Haozhong Wang (Ocean Univ. of China, no 238, Singling Rd., Songling, Laoshan Dist., Qingdao, China, coolice@ouc.edu.cn), and Yiqi Bai (Ocean Univ. of China, Qingdao, China)

In underwater acoustic communication systems, channel characteristics are mainly affected by changes in space and time. The changes embody the random fluctuations of the seabed and sea surface, and the effects of refraction and scattering caused by seawater layered media on the sound field. As the channel is time-varying and space-varying, the rapid fluctuation of communication signals, which is shown in the frequency selective fading in frequency-domain and the signal waveform distortion in the time domain, leads to a bad effect on the performance of the underwater acoustic communication system. The techniques to compensate or offset the effects of deep fading such as coding error correction and spatial diversity are widely used in underwater acoustic communication systems, which may cause a significant waste of limited communication efficiency. This paper analyzes the spatiotemporal fluctuation characteristics of both signal field and noise field, and summarizes the temporal and spatial variation rules. The effects of signal spatiotemporal fluctuation on communication systems are significantly reduced by reasonably selecting the communication signal parameters, equipment deployment depth and horizontal distance, which will guide the parameter configuration and network protocol optimization of communication systems.

**3:35**

**4pUW3. A modeling of synthetic aperture Image of a partially buried rigid cylinder on a flat boundary.** Kyungmin Baik (Acoust., Vib., Ultrasound Metrology Group, KRISS, Acoust., Vib., Ultrasound Metrology Group, Daejeon 34113, South Korea, kbaik@kriss.re.kr), Wan-Gu Kim (Acoust., Vib., Ultrasound Metrology Group, KRISS, Daejeon, South Korea), and Jungsoo Ryue (School of Naval Architecture and Ocean Eng., Univ. of Ulsan, Ulsan, South Korea)

Synthetic Aperture Sonar is a side-looking sonar that can obtain the equal resolution of the sonar images in lower frequencies than the conventional frequency ranges of Side Scan Sonar. The current study shows a part of the works being developed in the incorporation of the prototypes of Synthetic Aperture Sonar. As well as the conventional 2-D SAS, the system also has the bathymetric imaging in mind. In order to test the parameters and the algorithms imbedded in the system, a partially buried rigid cylinder was adopted as a simulation target for the broadside incident and the scattering amplitude by the cylinder was calculated by the Kirchhoff approximation. The features of monostatic scattering by a partially buried objects on the boundary are heavily dependent on its exposed surface out of the boundary [K. Baik and P. L. Marston, *IEEE J. Ocean. Eng.* **33**, 368 (2008)] due to the contributions by the flat boundary to the scattered direction. Delay of the reverberation between the boundary and the cylinder was considered by the inverse Fourier transform of total scattering form function [L. G. Zhang *et al.*, *J. Acoust. Soc. Am.* **91**, 1862 (1992)] in the modeling. Current study shows the effects the parameters of the scattering by the partially buried cylinder on the resulting simulated SAS images. [This work is supported by the project of "Development of towed interferometric Synthetic Aperture Sonar" (15-CM-SS-01).]

## OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday and Thursday. Most committee meetings will start at 4:30 p.m. EST, some committees choosing a later time as noted in the list below.

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:**

Acoustical Oceanography
Animal Bioacoustics
Architectural Acoustics
Engineering Acoustics
Physical Acoustics
Psychological and Physiological Acoustics
Signal Processing in Acoustics (6:30 p.m.)
Structural Acoustics and Vibration

**Committees meeting on Thursday are as follows:**

Biomedical Acoustics
Computational Acoustics
Musical Acoustics
Noise (6:00 p.m.)
Speech Communication
Underwater Acoustics

# Session 5aAAa

## Architectural Acoustics: Architectural Acoustics Potpourri II

David Manley, Chair
*DLR Group, 6457 Frances St., Omaha, NE 68106*

**Chair's Introduction—9:30**

## *Contributed Papers*

9:35

**5aAAa1. Small footprint head-related transfer function measurement apparatus for binaural heads utilizing one loudspeaker.** Zane T. Rusk (Dept. of Architectural Eng., The Penn State Univ., 104 Eng. Unit A, University Park, PA 16802, ztr4@psu.edu) and Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., University Park, PA)

Apparatuses for measuring head-related transfer-functions (HRTFs) often require large anechoic spaces with one or more arcs of loudspeakers. A measurement setup to obtain the HRTF of a Brüel & Kjær HATS 4100-D binaural head was designed and implemented that utilizes one loudspeaker in a 1309 ft$^3$ anechoic space with 18 in. wedges. By rotating the HATS along two axes, the characterization of many look directions can be achieved. The HATS can be rotated about the axis through the center of the ears via a turntable, and the yaw of the HATS can be adjusted via mounting gear on custom framing. HRTF measurements were completed using both maximum-length sequence and sine sweep signals in order to compare the two techniques. Comparisons between the measured 4100-D HRTF and the HRTF of a similar B&K binaural head will be discussed, as well as plans to use the 4100-D HRTF to compare binaural room impulse responses (RIRs) with spatial RIRs obtained with a spherical microphone array. [Work supported by a Graduate Assistance in Areas of National Need (GAANN) Fellowship.]

9:55

**5aAAa2. Panoptic reconstruction and dynamic synthesis of immersive virtual soundscapes.** Mincong Huang (Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, huangm5@rpi.edu), Samuel Chabot, Mallory Morgan, Hui Su (Rensselaer Polytechnic Inst., Troy, NY), and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

Visually generated spatial audio has been frequently explored in research on multimedia analysis. While these efforts have aimed at constructing congruent representation of recorded audiovisual experiences, it has yet to be directly applied to the rendering of virtual soundscapes. Moreover, few have dealt with scenarios in which audio information of captured physical environments is completely absent. This motivates the panoptic soundscape reconstruction and rendering technique presented in this work. The technique considers only captured panoramic visual environments with both embedded and annotated contextual metadata. The visual information is processed using existing pre-trained deep neural networks (DNNs) for semantic segmentation and object detection, with extracted information processed into sound objects. The embedded metadata is used to probabilistically analyze underlying contextual information related to the acoustic environments and generate predictions of its behaviors. Both sound objects and predictions are used as inputs for a spatial audio manager, which generates virtual sound sources and schedules them accordingly. The resulting soundscapes are rendered and experienced with room-centered virtual reality systems, such as Rensselaer's Collaborative-Research Augmented Immersive Virtual Environment Laboratory (CRAIVE-Lab), with potential adaptability to user-centered virtual reality devices. [Work supported by NSF No.1909229, Cognitive Immersive Systems Laboratory (CISL) and Army DURIP No. 68604-CS-RIP.]

10:15

**5aAAa3. Co-locating remote collaborators in immersive virtual environments using telematic systems.** Samuel Chabot (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, chabos2@rpi.edu), Jonathan Mathews (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY), Hui Su (Rensselaer Polytechnic Inst., Troy, NY), and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

The rapid shift to remote work has forever changed the dynamic of teams, which require the best tools to work collaboratively across any distance. Rensselaer is home to two immersive virtual environments, intelligent rooms with panoramic, human-scale projection screens, spatial audio loudspeaker arrays, and networks of time-of-flight and acoustical tracking sensors. This project seeks to "colocate" teams across both sites such that the experience mimics collaborating within the same room. Simple video conferencing software are rarely well-configured for large groups and do not consider users' spatial arrangements. This approach captures ultra-low-latency video and audio feeds of each space for presentation at either end, enabling a group at each location to communicate at 1-to-1 scale with the other. A spherical microphone array tracks multiple simultaneous speakers and adjusts their spatial positions across a Wave Field Synthesis loudspeaker array, maintaining audio-visual congruency. Users at each site can use the panoramic displays to present panoramic imagery or immersive data to be explored simultaneously by each group, facilitating a collaborative immersive experience, and interactions with the screen at one site may be echoed at the other to create the appearance of a single space. [Work supported by CISL, NSF No. 1229391, Army DURIP No. 68604-CS-RIP.]

10:35

**5aAAa4. Blind estimation of the direct-to-reverberant ratio using a beta distribution fit to binaural coherence.** Paul Calamia (Audio, Facebook Reality Labs, FRL, Redmond, WA 98052, pcalamia@fb.com), Nava Balsam, and Philip Robinson (Audio, Facebook Reality Labs, Redmond, WA)

Knowledge of the direct-to-reverberant ratio (DRR) between an acoustic source and a listening position or receiver can be useful for various acoustic and audio applications including dereverberation, source distance estimation, automatic speech recognition, and binaural synthesis. While the DRR can be computed easily from a room impulse response (RIR), blind estimation using acoustic sources of opportunity is necessary when such RIRs are not available. In this presentation, we describe an approach for blind estimation of the DRR which uses the magnitude-squared coherence (MSC) between the two channels of a binaural signal. The method involves fitting a beta distribution to the MSC values, aggregated over time and frequency, and deriving the estimate from the relationship between the DRR and the shape parameters of that distribution. Validation experiments utilizing speech convolved with binaural RIRs collected from a variety of publicly available datasets yield DRR estimates that are within the just-noticeable difference for DRRs in the range -15 to + 18 dB.

# Session 5aAAb

## Architectural Acoustics: Architectural Acoustics Potpourri III

Matthew T. Neal, Chair
*Otolaryngology and Communication Disorders, University of Louisville, 117 E Kentucky Street, Louisville, KY 40203*

**Chair's Introduction—11:15**

## *Contributed Papers*

**11:20**

**5aAAb1. Simulated acoustic characteristics of stage displacements in a shoebox hall: A case study in Cultural Centre Concert Hall in Taipei.** Yu-Chao Chen (Dept. of Architecture, National Taiwan Univ. of Sci. and Tech., 43 Keelung Rd., Sec. 4, Taipei 10607, Taiwan, d10813001@mail.ntust.edu.tw) and Lucky S. Tsaih (Dept. of Architecture, National Taiwan Univ. of Sci. and Tech., Taipei, Taiwan)

Since Berlin Philharmonic Hall opened in 1963, surround hall configuration has been preferred and used worldwide by architects for its dramatic visual appearance and the proximity to stage. However, according to Beranek's study, the top three concert halls in the world that have the best acoustics are in shoebox shape. To study the acoustic characteristics of stage displacement in a shoebox hall is the objective of this study. A simplified wired-frame model of Cultural Centre Concert Hall in Taipei is used as an example in this study. It is a stunning performing-arts building as mentioned by Beranek. Its seating capacity is 2077 and it is a shoebox hall. Five different stage displacements along the central axis have been used. The original ceiling configuration is used and moved with each stage displacements. Fixed materials and the relevant absorption coefficients are used in the study as well. Acoustic parameters such as music clarity, lateral refraction, early decay time as well as reverberation time (T30) are simulated. The results showed that for the best proximity, as stage and stage ceiling are moved toward the centre of shoebox hall, the simulated T30 could be increased to approximately 1.7 seconds in 1000 Hz. To meet the original T30 design goal in 2.25 s at 1000 Hz, about 600 $m^2$ of NRC 0.88 acoustic panel were used.

**11:40**

**5aAAb2. Concert hall preference: Investigating key factors related to overall average and individual preference.** Matthew T. Neal (Otolaryngol. and Commun. Disord., Univ. of Louisville, 117 E Kentucky St., Louisville, KY 40203, matthew.neal.2@louisville.edu) and Michelle C. Vigeant (The Pennsylvania State Univ., University Park, PA)

A holistic approach to study individual preference in concert hall acoustics is difficult to achieve. A careful balance must be struck between using well-controlled laboratory testing conditions and maintaining natural, realistic auralizations. To achieve this balance, measurements were made in 21 concert halls across North America and Europe using spherical-array processing techniques for spatially accurate auralizations, which included realistic instrument directivity. Measurements were made at 20 standard orchestral instrument locations at a seat 15 m from the stage. Using this database, a subjective study has been conducted, comparing 14 representative halls rated for preference and ten significant subjective terms from concert hall acoustics literature. Principle components and factor analyses identified three to four representative and interpretable factors to describe concert hall perception. Correlation analyses were then conducted between these key factors and the average preference to determine overall consensus factors of preference. Since individual preference is inherently undefined, correlation between each individual's preference ratings and these key factors and the repeatability of each subject were also investigated. Past literature suggests two distinct preference groups, but current results suggest the need for a continuous preference space, not represented by two groups and one variable alone. [Work supported by NSF Award 1302741.]

**12:00**

**5aAAb3. The just noticeable difference of early decay time (EDT).** Fernando M. del Solar Dorrego (Acoust., Penn State, 445 Waupelani Dr., Apt. B10, State College, PA 16801, fsolar@gmail.com) and Michelle C. Vigeant (Acoust., Penn State, University Park, PA)

The purpose of this study was to determine the just noticeable difference (JND) of early decay time (EDT) for broadband conditions and individual frequency bands. JNDs of room acoustic parameters are important for assessing if changes in geometry or materials in a room will have a discernible impact in the listener's perception. The EDT JND has been reported to be 5% in the ISO 3382-1:2009 standard, which is the same value usually quoted for the JND of classical reverberation time (T30). No formal studies have been undertaken so far to validate if this JND value is appropriate for EDT, or to determine its dependency on frequency. Measured spatial room impulse responses (SRIRs) from three concert halls were used as base cases, which were then modified with a computer algorithm to attain specified values of EDT. All stimuli were reproduced in an anechoic chamber with an array of 30 loudspeakers using third-order ambisonics. The method of constant stimuli, in which a psychometric function is fitted to the measured data, was used to determine the JND. Thirty subjects with musical backgrounds participated in the study. The obtained broadband JND was 26%, while individual frequency band JNDs were higher.

**12:20**

**5aAAb4. Acoustical design challenges of colossal public interiors.** Zühre Sü Gül (Architecture, Bilkent Univ., Ankara, Turkey, zuhre@bilkent.edu.tr), Mehmet Caliskan, and Zeynep Bora Ozyurt (Mezzo Studyo, Ltd., Ankara, Turkey)

In this study, a long-term experience with public buildings are discussed over architectural and acoustical parameters. Different codes and standards apply for a variety of public buildings in different countries. The criteria are generally set and can be satisfied for sound insulation metrics without an over-use of material. However, room acoustics design is always a challenge especially when the indoor space is excessive in scale, yet the criteria are also not always well-defined. In this contemporary era, the architectural styles emphasize even more the flexibility and the flow of spaces by connected volumes. This approach commonly ends up with linked circulation zones, huge foyers or almost monumental entrances. To exemplify and further discuss some of these cases, this study is framed around two buildings of different functions. The first case is Heydar Aliyev Center in Azerbaijan, with coupled inner galleries of circulation of museum and library zones and a grand foyer of the auditorium. Second case is the Kuwait International Airport Terminal possessing a grandeur of interior voids of check-in and concourse halls. For both cases the optimization of excessive reverberation is the major issue in a struggle of not overly applying acoustical materials. The selected indoor spaces at some locations demonstrate almost a free-field behavior. Accordingly, this study questions the use of common parameters, including reverberation time, strength and sound levels and the setting of proper criteria for such large public interiors.

## Session 5aAB

# Animal Bioacoustics: Passive Acoustic Monitoring of Animal Bioacoustic Signals

Selene Fregosi, Chair
*Oregon State University, 2030 SE Marine Science Drive, Newport, Newport, OR 97365*

**Chair's Introduction—9:30**

## *Contributed Papers*

**9:35**

**5aAB1. Tracking fin whales in a coastal fjord using passive acoustics.** Benjamin Hendricks (SoundSpace Analytics, 2845 Penrith Ave., Cumberland, BC V0R 1S0, Canada, benhendricks@soundspace-analytics.ca), Eric Keen, Janie Wray (North Coast Cetacean Society, Alert Bay, BC, Canada), Chenoah Shine (Dept. of Geography, Univ. of Victoria, Victoria, BC, Canada), Hussein M Alidina (Oceans, WWF-Canada, Victoria, BC, Canada), and Chris Picard (Gitga'at Oceans and Lands Dept., Hartley Bay, BC, Canada)

In the north-eastern Pacific, fin whales (*Balaenoptara physalus*) are typically found and studied in open waters near or seaward of the continental shelf. The Kitimat fjord system is a network of sounds and channels along the coast of British Columbia, Canada, and is one of only a few known fin whale habitats within confined coastal waters. In 2018, we deployed a network of hydrophones in Squally Channel, a proposed shipping lane located at the center of the fjord system, to monitor and track fin whales for a period of 2 years. The work is part of an effort to mitigate anthropogenic stressors on whales in the traditional territory of the Gitga'at First Nation. Here, we present spatial-temporal habitat use by fin whales in this confined inshore ecosystem based on more than 30 000 call detections and 7000 source localizations. Two distinct call types were recorded, a 20 Hz and a 40 Hz pulse, with distinctly different seasonal and diel trends. About 100 individual tracks were reconstructed from 20 Hz call sequences and analyzed for calling characteristics and movement patterns. A better understanding of fin whale habitat in the north-east Pacific is of special importance in light of an active discussion about the conservation status of this large baleen whale in Canadian waters.

**9:55**

**5aAB2. Acoustic ecological investigation of estuary habitat on Indo-Pacific humpback dolphin (*Sousa chinensis*) in Yunlin, Taiwan.** Wei-Chun Hu (Eng. Sci. and Ocean Eng., National Taiwan Univ., No. 130 Keelung Rd., Section#3, Rm. 207, Taipei 106, Taiwan, william_hu@outlook.com), Chi-Fang Chen (Eng. Sci. and Ocean Eng., National Taiwan Univ., Taipei, Taiwan), and Lien-Siang Chou (Inst. of Ecology and Evolutionary Biology, National Taiwan Univ., Taipei, Taiwan)

A sunspecies/population of Indo-Pacific humpback dolphins (IPHD, *Sousa chinensis*) lives in the shallow waters of Eastern Taiwan Strait. Their habitat is very close to the coast and suffering from the impact of human activities and marine pollution; the survival of this vulnerable group is at high risk. The long-term visual survey result shows that the Xinhuwei estuary has been an IPHD hot spot. Two Passive acoustic monitoring (PAM) stations with temperature sensors were deployed to detect IPHD sounds in the estuary habitat. During the past four seasons, underwater marine recorders and temperature-depth data loggers were deployed at two locations in the estuary of Xinhuwei creek, Yunlin, Taiwan. The total duration of valid data for each station is at least 19 consecutive days during each season. IPHD's click-trains and whistles were counted by a supervised detection method. The results show that the trend of click-trains (echolocation and foraging sounds) is positively correlated with sea temperature. The most active foraging behavior occurs during the daytime of summer. Whistles (social and communication sounds) are most common during spring. Except in winter, the number of click-trains and whistles are larger at the deeper measuring station (water depth of 11 m). This research was funded by Formosa Petrochemical Corporation.

**10:15**

**5aAB3. Near real time passive acoustic monitoring in the Santa Barbara Channel.** Megan Wood (Marine Biology, Texas A&M Univ. at Galveston, PO Box 1675, Bldg 3029 Rm. 130, Galveston, TX 77553, mwood@tamu.edu), Mark Baumgartner (Biology, Woods Hole Oceanographic Inst., Woods Hole, MA), Morgan Visalli (Univ. of California Santa Barbara, Santa Barbara, CA), and Ana Širović (Marine Biology, Texas A&M Univ. at Galveston, Galveston, TX)

The coastal waters off Southern California are feeding grounds for a variety of baleen whale species including blue whales, fin whales, and humpback whales. The feeding grounds overlap with highly used shipping lanes in the Santa Barbara Channel (SBC), increasing the risk of ship-whale interactions. In November 2019, a buoy equipped with a digital acoustic monitoring (DMON) instrument was deployed in the SBC and has been actively recording sounds and transmitting detection information in near real time since then. Data are transmitted in fifteen-minute summary periods of detected pitch tracks, which are reviewed by an analyst onshore for blue, fin, and humpback whale calls. Humpback whales were the most prolific callers, with detection rates reaching 80% of daily summary periods, whereas fin and blue whales rarely exceeded 20%. Baleen whale detections were low in late fall, with humpback whale detections increasing throughout winter and in spring and blue whale detections peaking during the summer. As part of the Benioff Ocean Initiative project to reduce whale strikes, these data are combined with visual surveys in the SBC and habitat models to produce a whale presence metric. This metric can be used by regional stakeholders to guide policy decisions and enhance conservation efforts.

**10:35**

**5aAB4. Detection probability and density estimation of fin whales by a seaglider.** Selene Fregosi (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, selene.fregosi@gmail.com), Danielle V. Harris (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St Andrews, St Andrews, United Kingdom), Haruyoshi Matsumoto, David K. Mellinger (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., Newport, OR), Stephen W. Martin, Brian Matsuyama (National Marine Mammal Foundation, San Diego, CA), Jay Barlow (NOAA National Marine Fisheries Service, Southwest Fisheries Sci. Ctr., Marine Mammal and Turtle Div., La Jolla, CA), and Holger Klinck (Ctr. for Conservation Bioacoustics, Cornell Lab of Ornithology, Cornell Univ., Ithaca, NY)

The effectiveness of gliders is evaluated for passive acoustic density estimation of fin whales. An estimate of the probability of detection as a

5a FRI. AM

function of range (or, equivalently, effective survey area) is required to estimate density from acoustic data collected by a single-hydrophone glider platform. A cabled hydrophone array was used to estimate fin whale localizations and tracks concurrently with a glider survey. Fin whale tracks were used as detection trials and a detection function for snapshots containing fin whale 20 Hz pulses recorded by the glider was modeled using a generalized additive model. Detection probability was strongly dependent on 40 Hz noise levels recorded on the glider. At the median noise level of 97 dB *re* 1 $\mu Pa^2/Hz$, detection probability was near one at zero horizontal distance, and maximum detection range was near 40 km. The estimated effective survey area at this noise level was 870 $km^2$. Using estimates of vocal rates and group size from tagged and tracked fin whales, respectively, the density was estimated as 2.4 fin whales per 1000 $km^2$ (coefficient of variation 0.55). The framework presented here could be applied to other baleen whale species to advance the use of gliders for density estimation of cetaceans.

FRIDAY MORNING, 11 DECEMBER 2020 — 9:30 A.M. TO 11:00 A.M.

# Session 5aBAa

## Biomedical Acoustics: General Biomedical Acoustics: Therapeutics I

John M. Cormack, Chair
*Department of Medicine, University of Pittsburgh, Pittsburgh, PA 15261*

**Chair's Introduction—9:30**

### *Contributed Papers*

**9:35**

**5aBAa1. Spatial variation of ultrasonic attenuation and speed of sound in brain tissue visualized by parametric imaging.** Will R. Newman (Phys., Rhodes College, Memphis, TN, newwr-21@rhodes.edu), Cecille Labuda (Phys. and Astronomy, Univ. of MS, University, MS), and Brent K. Hoffmeister (Phys., Rhodes College, Memphis, TN)

The ultrasonic properties of brain have been explored to a limited extent, however the spatial variation of these properties is not well characterized. The goal of this study was to measure the speed of sound and the frequency slope of attenuation in brain tissue at multiple locations to generate parametric images that characterize their spatial distribution. Tissue specimens were 1-cm thick slices of preserved sheep brain prepared from the coronal, sagittal and transverse anatomic planes. Ultrasonic measurements were performed using broadband transducers with center frequencies of 3.5, 5.0, 7.5, and 10 MHz. The transducers were mechanically scanned to acquire signals from all locations on each slice. Structures visible in the parametric images were consistent with the known morphologic features of the brain. White matter and gray matter appeared to be distinguishable in most images. Measured values for the spatial mean and standard deviation of the frequency slope of attenuation ranged between 0.723–1.06 and 0.194–0.501 $dBcm^{-1}MHz^{-1}$, respectively, depending on the tissue slice and transducer frequency. Measured values for the spatial mean and standard deviation of the speed of sound ranged between 1520–560 and 6–15 $ms^{-1}$, respectively. Spatial variation of these properties were clearly visualized in the parametric images.

**9:55**

**5aBAa2. Parametric imaging of ultrasonic backscatter of fixed sheep brain.** Cecille Labuda (Phys. and Astronomy, Univ. of MS, 108 Lewis Hall, University of MS, University, MS 38677, cpembert@olemiss.edu), Will R. Newman, and Brent K. Hoffmeister (Rhodes College, Memphis, TN)

Ultrasound backscatter properties of aggregate brain tissue are available in the literature, however, reporting on the spatial variation of backscatter over tissue volumes is limited. The spatial variation of the apparent integrated backscatter (AIB) and the logarithmic backscatter amplitude decay constant (BADCL) was characterized for 1-cm thick samples from the coronal, sagittal and transverse anatomic planes of fixed sheep brain. Submerged samples were exposed to ultrasound using broadband transducers with center frequencies of 3.5, 5.0, 7.5, and 10 MHz by scanning over the samples in half-beamwidth step sizes to measure the backscattered signal. Parametric images of the AIB and BADCL showed a clear correlation between morphological features in the tissue samples and the images with the AIB giving the better representation of the overall tissue structure. Over all samples and frequencies, the range of the spatial mean of the AIB was −77.7 to −59.8 dB with the standard deviation ranging from 3.14 to 6.99 dB. The spatial mean of the BADCL ranged from 0.0350 to 0.152 $\mu s^{-1}$ with standard deviations ranging from 0.0820 to 0.111 $\mu s^{-1}$. For both parameters, morphological features of the tissue become more distinctive with increasing frequency.

**10:15**

**5aBAa3. Monitoring of atrial ablation using cyclic variation of integrated backscatter.** Scott Anjewierden (Dept. of Biomedical Eng., Cleveland Clinic Lerner Res. Inst., 9500 Euclid Ave., Cleveland, OH 44195, sanjewierden@gmail.com), Oussama M. Wazni (Dept. of Cardiovascular Medicine, Cleveland Clinic Foundation, Cleveland, OH), D. G. Vince (Dept. of Biomedical Eng., Cleveland Clinic Lerner Res. Inst., Cleveland, OH), Mohamed Kanj, Walid Saliba (Dept. of Cardiovascular Medicine, Cleveland Clinic Foundation, Cleveland, OH), and Russell J. Fedewa (Dept. of Biomedical Eng., Cleveland Clinic Lerner Res. Inst., Cleveland, OH)

Treatment of atrial fibrillation (AF) often relies on radiofrequency ablation (RFA) of atrial tissue. However, current treatment is associated with a >20% recurrence rate, in part due to inadequate monitoring of tissue viability during ablation. Previous work has utilized cyclic variation of integrated backscatter (CVIB) as an early indicator of myocardial recovery from ischemia. Our aim was to demonstrate the use of CVIB to distinguish normal and ablated myocardium. An AcuNav 10F catheter was used to collect radiofrequency signals from the posterior wall of the left atrium of patients before and after RFA for AF. The normalized power spectrum was obtained and integrated backscatter (IB) was extracted across two continuous heart cycles to calculate the average CVIB. Data from 14 patients demonstrated a significant difference in the magnitude of the CVIB before and after ablation (9.0 vs 6.0 dB, $p < 0.001$). However, no significant changes were identified when evaluating ECG-gated IB values before and after ablation: ventricular end-systole (−9.8 vs −8.6, p = 0.24), atrial end-diastole (−8.2 vs −8.6, p = 0.72) or ventricular end-diastole (−7.7 vs −8.9, p = 0.18). CVIB is able to differentiate normal and ablated myocardium and may be useful in monitoring atrial ablations.

**10:35**

**5aBAa4. Real-time control of radiofrequency ablation by three-dimensional echo decorrelation imaging.** Peter D. Grimm (Biomedical Eng., Univ. of Cincinnati, 3498 Tarpis Ave., Cincinnati, OH 45208, grimmpd@mail.uc.edu), Elmira Ghahramani Z., E. G. Sunethra Dayavansha, Michael Swearengen, and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH)

Feasibility for real-time control of radiofrequency ablation (RFA) by three-dimensional (3-D) echo decorrelation imaging, with potential application to liver cancer treatment, is investigated. *Ex vivo* bovine liver is treated using a clinical RFA system (RITA 1500X generator with StarBurst probe, Angiodynamics). During ablation, pairs of echo volumes from a Siemens SC2000 scanner with a 4Z1c matrix array probe (inter-frame time 18 ms) are recorded as in-phase and quadrature (IQ) data. 3-D echo decorrelation maps are computed from these volume pairs every 11 s. Cumulative echo decorrelation, defined as the temporal maximum decorrelation for each voxel, is spatially averaged within a spherical region of interest (ROI) centered at the RF needle tip. Once the spatially averaged cumulative decorrelation within the spherical ROI exceeds a predefined threshold, ablation is halted by communication with the RF generator via a pneumatic switch connected to a microcontroller circuit. The control threshold and ROI were determined by empirically evaluating predictive performance of decorrelation in preliminary uncontrolled ablation trials. Controlled and uncontrolled ablation are compared based on correspondence of sectioned tissue histology to targeted ablation zones, receiver operating characteristic curve analysis of local ablation prediction by echo decorrelation, treatment duration, and ablation rate.

FRIDAY MORNING, 11 DECEMBER 2020 — 11:15 A.M. TO 12:45 P.M.

# Session 5aBAb

## Biomedical Acoustics: General Biomedical Acoustics: Therapeutics II

John M. Cormack, Chair
*Department of Medicine, University of Pittsburgh, Pittsburgh, PA 15261*

**Chair's Introduction—11:15**

### *Contributed Papers*

**11:20**

**5aBAb1. Growth and regeneration of sympathetic neuron axons induced by pulsed low intensity ultrasound.** Jeannette Nyiramana (Biomedical Eng., Tulane Univ., 6823 St Charles Ave., New Orleans, LA 70118, jnyiramana@tulane.edu), Kendall Walker (Neurosci., Tulane Univ., New Orleans, LA), and Damir Khismatullin (Biomedical Eng., Tulane Univ., New Orleans, LA)

Over a quarter million Americans are currently living with spinal cord injuries (SCIs), costing $3 billion a year to manage [1]. SCI results in myelin loss, axonal degradation, and death of sympathetic neurons, leading to lesion formation and disruption of the primary communication pathway in the spinal cord. Low intensity ultrasound has been shown to improve axonal regeneration in the peripheral nervous system, but its effect on cells in the central nervous systems is not yet studied. The objective of this work is to investigate the growth-stimulating effects of low to mid-intensity ultrasound on axons of sympathetic neurons. To find optimal ultrasound intensities, neuroendocrine-type PC-12 cells were cultured and treated with pulsed ultrasound at varying intensities. Cells were cultured in deprived states to model a post-SCI environment, while being treated with neurotrophic factor (NGF) to induce their neuron differentiation. Images of cells were analyzed for number of cells and neurite outgrowth. Experiments with PC-12 cells found that an ultrasound intensity of 1.02 W/cm$^2$ led to optimal neurite outgrowth when analyzing the rate of neurite outgrowth and total neurite outgrowth at the final day of experimentation. Our ultimate goal is to develop an effective, noninvasive SCI treatment based on the ultrasound technology.

**11:40**

**5aBAb2. Histological effects on *ex vivo* rat tendon from focused ultrasound treatment.** Molly Smallcomb (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, molly.smallcomb@gmail.com), Jacob C. Elliott (Graduate Program in Acoust., Penn State Univ., State College, PA), Sujata Khandare, Ali A. Butt, Meghan E. Vidt (Biomedical Eng., Penn State Univ., State College, PA), and Julianna Simon (Graduate Program in Acoust., Penn State Univ., State College, PA)

Focused ultrasound (fUS) therapy can induce controllable mechanical damage through bubble creation, oscillation, and collapse. However, highly collagenous tissues like tendon are resistant to mechanical fractionation with fUS. Our objective is to histologically evaluate whether fUS-induced mechanical disruption is achievable in rat tendon. *Ex vivo* Achilles (AT) and supraspinatus (ST) tendons were exposed to 1.5 MHz pulses of 0.1–10 ms repeated at 1–100 Hz for 15–60 s with peak pressures $p_+ = 69$ MPa, $p_- = 24$ MPa. B-mode ultrasound was used to monitor hyperechogencity during treatment. Samples were stained with Hematoxylin and Eosin (H&E) or alpha-nicotinamide dinucleotide diaphorase (αNADH-d). Results showed

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successful bubble creation for all samples; however, all samples did not show histological injury. Samples treated with 10-ms pulses at 1 Hz for 15 s displayed only slight disruption (2/5 AT, 2/5 ST). When treatment time increased to 30 s, thermal injury dominated over mechanical effects. Shorter pulse lengths (1 ms at 10 Hz; 0.1 ms at 100 Hz) resulted in localized fiber separation (5/10 AT, 2/10 ST). Future work will investigate how fUS influences mechanical properties of tendon and whether it can induce a healing response *in vivo.* [Work supported by NIH NIBIB EB027886; NSF GRFP DGE1255832 (Smallcomb).]

**12:00**

**5aBAb3. Therapeutic effects of ultrasound on dermal wound healing in diabetic mice.** Melinda A. Vander Horst (Biomedical Eng., Univ. of Rochester, 204 Robert B. Goergen Hall, Rochester, NY 14627, mvander7@ur.rochester.edu), Carol H. Raeman, Diane Dalecki (Biomedical Eng., Univ. of Rochester, Rochester, NY), and Denise C. Hocking (Pharmacology and Physiol., Univ. of Rochester, Rochester, NY)

Some evidence indicates that relatively low intensity, pulsed ultrasound can enhance soft tissue regeneration in both animal models and humans. However, the effectiveness of therapeutic ultrasound can vary among populations. Here, we investigated effects of ultrasound on dermal wound healing using a murine model of chronic, diabetic wounds. Full thickness, punch biopsy wounds were made on the dorsal skin of genetically diabetic, male mice. Wounds were exposed to 1-MHz pulsed ultrasound (2 ms pulse, 100 Hz PRF, 0–0.4 MPa) for 8 min per day, for either 2 or 3 weeks. After 2 weeks, the average granulation tissue thickness at the center of wounds exposed to 0.4 MPa ultrasound was significantly increased compared to sham-exposed wounds. Interestingly, the mean granulation tissue thickness of one subpopulation of mice exposed to 0.4 MPa was significantly greater than that of a second subpopulation ($594 \pm 41$ $\mu$m, n = 9 vs $62 \pm 25$ $\mu$m, n = 10). Increasing the treatment time to 3 weeks increased the percentage of mice with fully re-epithelialized wounds from 47% at 2-weeks to 86%, and significantly reduced wound diameter. These data demonstrate a dose-dependent difference in the responsiveness of young, male diabetic mice to pulsed ultrasound treatment protocols.

**12:20**

**5aBAb4. Microwave-induced thermacoustic signal characteristics during pulsed microwave ablation.** Audrey L. Evans (Elec. and Comput. Eng., Univ. of Wisconsin-Madison, 1415 Eng. Dr. RM 3415, Madison, WI 53706, alevans3@wisc.edu), Susan C. Hagness, and Chu Ma (Elec. and Comput. Eng., Univ. of Wisconsin-Madison, Madison, WI)

Microwave ablation (MWA) is a minimally invasive thermal therapy in which an interstitial antenna delivers microwave power locally to diseased tissue to cause cell death. Thermoacoustic (TA) signals can be generated with internally pulsed microwave energy that simultaneously ablates tissue. When pulsed microwave energy is absorbed by tissue, the tissue undergoes an incremental temperature rise which leads to thermoelastic expansion and the generation of an acoustic wave that can be detected by ultrasound transducers. The characteristics of the TA signals are linked to features of the local ablation environment from which they are generated, and thus can be exploited for ablation monitoring. In this experimental and computational study, we examine the fundamental characteristics of TA signals generated from within the ablation zone and observe how the TA signal changes as the ablation zone evolves. We performed pulsed microwave heating experiments in non-coagulating and coagulating liquids and measured the resulting TA signals. We also performed multiphysics simulations of the MWA process that encompass the electromagnetic, thermal, and acoustics physics. This study provides insights into TA signal generation and propagation during MWA and establishes the feasibility of using the TA signals for ablation monitoring.

## Session 5aEA

## Engineering Acoustics: General Topics in Engineering Acoustics IV

Thomas E. Blanford, Cochair
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Caleb F. Sieck, Cochair
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**Chair's Introduction—9:30**

### *Contributed Papers*

**9:35**

**5aEA1. Influence of phononic material filters on nonlinear ultrasound measurements.** Elizabeth Smith (Mech. Sci. and Eng., Univ. of Illinois at Urbana Champaign, Mech. Eng. Bldg., 1206 W Green St., Urbana, IL 61801, esmith19@illinois.edu) and Kathryn Matlack (Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Nonlinear ultrasound (NLU) is a nondestructive evaluation method that is sensitive to microscopic material damage, which is highly desirable information to prevent failure of critical components. The damage is correlated to the second harmonic generated by a sinusoidal signal as it propagates through a material. However, measurements of the second harmonic are plagued by experimentally induced nonlinearities and require careful calibrations that have limited them to laboratory measurements. Here, we propose the use of phononic materials with ultrasonic filtering properties to reduce extraneous nonlinearities. Finite element simulations were used to design phononic materials to transmit an ultrasonic wave but forbid the propagation of its second harmonic. Phononic filters were fabricated with additive manufacturing and experimentally characterized in the ultrasonic regime. Results show that the phononic materials behave as low-pass filters, where the cut-off frequency is controlled by the phononic material's unit cell geometry. Filters were then incorporated into NLU measurements. Experimental results show that the phononic filter decreases the measured second harmonic amplitude and removes extraneous nonlinearities from the transducer, therefore better isolating second harmonic generation in a material. This work suggests that phononic materials could enable in situ NLU measurements and of critical components.

**9:55**

**5aEA2. Simultaneous transmit and receive for self-sensing in ultrasonic measurement systems.** Matthew S. Byrne (Elec. and Comput. Eng., Univ. of Texas at Austin, 2501 Speedway, Rm. 5.838-A (Seat 5), Austin, TX 78751, mbyrne@utexas.edu), Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Andrea Alu (Photonics Initiative, CUNY Adv. Sci. Res. Ctr., New York, NY)

Simultaneous transmit and receive (STAR) systems have recently enabled sending and receiving of radio-frequency (RF) signals at the same time and at the same frequency using a single antenna. This has led to commercialization efforts with the promise of doubling the throughput of traditional radio systems, including Wi-Fi and future 5G cellular communications. Prior to these advances, researchers in vibration control have been developing self-sensing actuator systems, also referred to as sensoriactuators or sensorless control systems. Inspired by these developments, we combine and extend these approaches to explore the development of STAR functionality in an acoustic measurement system at ultrasonic frequencies. We apply self-interference cancellation (SIC) to a time domain measurement in order to demonstrate the potential for a practical, single-transducer ultrasonic nondestructive evaluation (NDE) system to measure continuous echo returns while it is actively transmitting at the same frequency. Theoretical models and preliminary experimental results will be presented and discussed.

**10:15**

**5aEA3. Numerical study of an acoustic leaky wave antenna.** Omar A. Bustamante (Grupo de Acústica y Vibraciones, Instituto de Ciencias Aplicadas y Tecnología, Universidad Nacional Autónoma de México, Circuito Exterior S/N, Ciudad Universitaria, Mexico City 04510, Mexico, omarb.0502@gmail.com), Eduardo Romero-Vivas (Grupo de Investigación en Acústica y Procesamiento de Señales, Centro de Investigaciones Biológicas del Noroeste, S.C., CIBNOR, La Paz, Baja California Sur, Mexico), and Roberto Velasco-Segura (Grupo de Acústica y Vibraciones, Instituto de Ciencias Aplicadas y Tecnología, Universidad Nacional Autónoma de México, Ciudad Universitaria, Mexico City, Mexico)

Acoustic leaky wave antennas (LWAs), introduced as frequency scanning devices, have been studied in recent years for Direction of Arrival (DOA) and highly directional frequency-selective transmission due to their advantages over multi element arrays such as compactness and low-power requirements. This work presents a numerical study through Finite Element Method (FEM) that solves the Helmholtz Equation using code based on the open-source FEniCS Project. The model under study is a one-dimensional acoustic LWA with axysimetric open channels that performs as a transmitter from broadside to endfire directions. We describe the numerical method applied and validate the code with a lumped-element analysis approximation over the frequency range of interest, showing good agreement in the results. Visualization of different aspects of the LWA like the dispersion relation and scattering parameters, among others are presented. Discussion also focus on the scripting possibilities for automation and inspection of large parameter spaces. This approach proves useful when considering termination effects like end-correction or when exploring new designs with irregular geometries that could provide better performing, but where theoretical modeling turns complicated. [Work supported by UNAM-PAPIIT TA100620.]

10:35

**5aEA4. An alternative approach to window ANC.** Stephen C. Thompson (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, sct12@psu.edu) and Andrew Dittberner (GN Hearing, Glenview, IL)

Outdoor noise can enter an office workspace through an open window. An active noise Control system Is described that can provide global reduction in the workspace of noise from outside the window. The system uses (1) a microphone array outside the window to detect the magnitude and direction of incoming sounds; (2) a sparse array of loudspeakers in and around the window to radiate sound into the office space to cancel the entering outside sound, and (3) a processing system to derive appropriate signals from the microphone array to drive the speaker array to achieve cancellation. The sparse array leaves the window mostly open and unobstructed. Array shading coefficients that match the radiation pattern of the array to that of the window aperture can be derived in advance. ANC processing on each outdoor noise source provides the signals to drive the cancellation array. A feedforward system built to this design concept should provide reasonable cancellation with minimal window obstruction and fewer optimization channels than a more conventional window cancellation system.

FRIDAY MORNING, 11 DECEMBER 2020 9:30 A.M. TO 11:00 A.M.

## Session 5aNSa

## Noise and Psychological and Physiological Acoustics: Advances in Hearing Protection Devices I

Cameron J. Fackler, Cochair
*3M Personal Safety Division, 7911 Zionsville Rd., Indianapolis, IN 46268*

William J. Murphy, Cochair
*Division of Field Studies and Engineering, Noise and Bioacoustics Team, National Institute for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998*

Elliott H. Berger, Cochair
*Berger Acoustical Consulting, 221 Olde Mill Cove, Indianapolis, IN 46260*

**Chair's Introduction—9:30**

### *Invited Papers*

9:35

**5aNSa1. Towards a metatheory explaining hearing protection device use among workers.** Olivier Doutres (Mech. Eng., ETS (Ecole de Technologie Superieure), Montreal, QC, Canada, olivier.doutres@etsmtl.ca), Jonathan Terroir (INRS, Vandoeuvre-lès-Nancy, France), Caroline Jolly, Chantal Gauvin (IRSST, Montreal, QC, Canada), Laurence Martin (Faculty of Medicine, Université de Montréal, Montreal, QC, Canada), and Alessia Negrini (IRSST, Montreal, QC, Canada)

Offering hearing protection devices (HPDs) to workers exposed to hazardous noise is a common noise control strategy for preventing noise-induced hearing loss (NIHL). However, HPDs are not always worn, or are used incorrectly and inconsistently by workers, which explains their limited efficiency. Numerous models based on social cognition theories aim to improve hearing protection training programs and to increase HPD use. They identify significant factors associated with worker's behaviors regarding HPD use (e.g., perceived barriers and benefits to HPD use, perceived self-efficacy). However, these models do not detail (dis)comfort aspects originating from complex interactions between physical characteristics of the triad "user/HPD/work environment" while these aspects are known to largely influence HPD (mis)use. The objective of this communication is to propose a metatheory explaining HPD (mis)use and based on the integration of an HPD comfort model into an existing social cognition model already developed for HPDs. This holistic description of the HPD use, involving both physical and psychosocial characteristics of the triad, could be used as a tool for the stakeholders (e.g., researchers, manufacturers, preventers) involved in the protection of workers from NIHL.

**9:55**

**5aNSa2. Speech intelligibility tests for hearing impaired persons using hearing protectors.** Sandra Dantscher (Inst. for Occupational Safety and Health of DGUV, Alte Heerstrasse 111, Sankt Augustin 53757, Germany, sandra.dantscher@dguv.de) and Peter Sickert (Laerm- und Gehoerschutz-Consult Peter Sickert, Nuremberg, Germany)

The combination of hearing impairment and the requirement to use hearing protection can create severe problems regarding situational awareness, signal audibility and communication ability. Nevertheless, for persons with a noise induced hearing loss a reliable protection against harmful noise is especially important. The German Social Accident Insurance (DGUV) founds a project that investigates a number of approaches to protect the vulnerable group of persons with a hearing loss at noisy workplaces, providing at the same time sufficient communication ability. In Germany, signal audibility with hearing protectors is addressed via a calculation using the sound attenuation of the hearing protector and a loudness model. But speech intelligibility, so far, cannot be quantified, neither by calculation nor standardized tests. The DGUV project investigates different types of speech tests respectively speech audiometry in the lab and the field. One aim is to quantify the effect of different types of hearing protectors (active and passive), another one to assess the feasibility of the different test approaches. We present results for simple, work-related messages in comparison to everyday sentences and a matrix sentence test. The effect of different hearing protectors is analysed in relation to the amount of hearing loss.

**10:15**

**5aNSa3. High fidelity hearing protection devices: Attenuation in human ears and manikin devices.** Colleen Le Prell (Univ. of Texas at Dallas-Callier Ctr., 1966 Inwood Rd., Callier Ctr., Rm. J216, Dallas, TX 75235, colleen.leprell@utdallas.edu) and Tess Zaccardi (Univ. of Texas at Dallas-Callier Ctr., Dallas, TX)

Flatness of attenuation was measured for high-fidelity hearing protection devices (HPDs) using microphone-in-real-ear (MIRE) and behavioral real-ear-attenuation-at-threshold (REAT) protocols. In addition, participants completed the words-in-noise (WIN) test with and without HPDs, and provided subjective ratings of perceived sound quality via surveys administered after listening to music with and without HPDs. Attenuation was also measured using a manikin device, with both the manikin-based microphone and the MIRE probe tube used to make measurements inside the artificial ear coupler. Among human participants, statistically significant correlations revealed decreased WIN performance and poorer music quality ratings as achieved HPD attenuation increased. There was also a statistically significant correlation in which shallower slope (flatter attenuation) was correlated with higher perceived music quality ratings. Achieved attenuation decreased in the coupler when the probe tube was inserted into the manikin, suggesting HPD seals were compromised by the probe tube. Consistent with the manikin data, attenuation in human ears was greater when measured using REAT, suggesting similar compromise of the HPD seal during MIRE tests. To assure high fidelity HPDs are well fit, REAT testing is recommended to verify both overall attenuation and flatness of attenuation in individual listeners.

**10:35**

**5aNSa4. Earcanal anthropometry analysis for the design of realistic artificial ears.** Bastien Poissenot-Arrigoni (Mech. Eng., ETS (Ecole de Technologie Superieure), 1100 Rue Notre-Dame Ouest, Montréal, QC H3C 1K3, Canada, bastien.poissenot.1@ens.etsmtl.ca), Chun Hong Law, Franck Sgard (IRSST, Montreal, QC, Canada), and Olivier Doutres (Mech. Eng., ETS (Ecole de tTechnologie Superieure), Montreal, QC, Canada)

How in-ear devices fit in the ear strongly influences the acoustical and mechanical (dis)comforts induced to the wearer. As important variations in the ear geometry exist based on gender, age and ethnicity, several studies collected ear anthropometric data as a basis for designing ear-mounted products. However, most of these studies focused on the ergonomic design of earbuds, and are thus limited to the geometry of the pinna, and concha (where earbuds fit). Few studies explored geometrical earcanal data for the design of intra-auricular hearing protectors that fit up to the earcanal second bend. The design of earplugs that fit to the widest range of earcanals requires realistic acoustical test fixtures representative of the population. This study uses statistical analysis and artificial intelligence based algorithms to cluster 32 Canadian workers pairs of earmolds scans as a function of earcanal curvilinear axis length, entrance, first and second bend cross sections area and circumference, but also earcanal tortuosity and cross sections aspect ratios. Dimensions relevant to cluster earcanals will be collected on a hundred of ears in a future study to design artificial ears which capture the inter-individual variability in mechanical and acoustical objective indicators related to the most important earplugs (dis)comfort attributes.

## Session 5aNSb

# Noise and Psychological and Physiological Acoustics: Advances in Hearing Protection Devices II

Cameron J. Fackler, Cochair
*3M Personal Safety Division, 7911 Zionsville Rd., Indianapolis, IN 46268*

William J. Murphy, Cochair
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**Chair's Introduction—11:15**

### *Invited Papers*

**11:20**

**5aNSb1. An explanation of the decrease in the earplug noise reduction when combined with an earmuff on an acoustic test fixture.** Yu Luan (Dept. of Mech. Eng., École de Technologie Supérieure, 1100 Rue Notre-Dame Ouest, Montreal, QC H3C 1K3, Canada, yu.luan.1@ens.etsmtl.ca), Olivier Doutres (Dept. of Mech. Eng., École de Technologie Supérieure, Montreal, QC, Canada), Hugues Nélisse, and Franck Sgard (Institut de Recherche Robert-Sauve´en Sante´et en Se´curite´du Travail, Montreal, QC, Canada)

For people working in high noise level environments, typically above 105 dBA, double hearing protection (DHP) systems (earplugs worn in combination with earmuffs) are always recommended. However, it is very difficult to predict the DHP overall sound attenuation due to the complexity of the system. A recent experimental study has shown on human subjects and an acoustic test fixture (ATF) that the noise reduction (NR) of the earplug decreases considerably when an earmuff is worn over it but the reason is still not fully understood. In this work, specially designed experiments using an ATF are proposed in order to explain this observation. The focus is put on the NR of the single earplug and that of the earplug in the DHP system. The respective effects on the DHP attenuation of various sound transmission paths involved in the system (both air-borne and structure-borne) are analyzed by modifying the system coupling conditions or controlling the sound pressure level under the earmuff. This work will help better understand the interactions between the system components and will provide the grounds for developing a prospective numerical model to predict the sound attenuation of a DHP system.

**11:40**

**5aNSb2. A three-dimensional finite-element model of a human head for predicting the objective occlusion effect induced by earplugs.** Huiyang Xu (Ecole de technologie superieure, 1100 Rue Notre-Dame Ouest, Montreal, QC H3C 1K3, Canada, huiyang.xu.1@ens.etsmtl.ca), Franck Sgard (IRSST, Montreal, QC, Canada), Kévin Carillo, Eric Wagnac, and Jacques De Guise (Ecole de Technologie Superieure, Montreal, QC, Canada)

In a noisy environment, wearing a correctly fitted earplug is the sine qua non condition to prevent noise-induced hearing loss. However, this condition is often unfulfilled due to the discomforts induced by the wearing of the earplug, among which the acoustical discomfort is influenced by the occlusion effect. Objectively, this phenomenon is quantified by a low frequency amplification of the sound pressure in the earcanal, induced by bone-conduction when the earcanal is occluded. Numerical models can go beyond the practical and ethical limits of experiments on living humans. Thus they can be an efficient and helpful tool to evaluate the occlusion effect. They also make it possible to better understand its underlying physical mechanisms associated with different factors concerning the anatomy, earplug and stimulation and ultimately how to reduce it. Thereby, a three-dimensional finite-element model of a human head is developed to compute the occlusion effect induced by earplug under a bone-conducted stimulation. Good agreement is obtained between the simulation results and the experimental data available in the literature giving confidence in the model to predict the occlusion effect. The model is exploited to investigate the individual effects of various factors (e.g., earplug and tissue properties) on the occlusion effect.

12:00

**5aNSb3. A finite element model to study the earplug contribution to the objective occlusion effect.** Kévin Carillo (École de Technologie Supérieure, 1100 Rue Notre-Dame Ouest, Montréal, QC H3C 1K3, Canada, kevin.carillo.1@ens.etsmtl.ca), Olivier Doutres (Mech. Eng., ETS (Ecole de tTchnologie Superieure), Montreal, QC, Canada), and Franck Sgard (IRSST, Montreal, QC, Canada)

The use of earplugs is commonly associated with an amplification of low frequency physiological noises in the earcanal referred to as the occlusion effect. In the literature, the type of earplug has been shown to significantly influence the occlusion effect in particular when inserted deeply enough. However, few studies have investigated the physical mechanisms that rule the earplug contribution. Their understanding is necessary to ultimately act on the earplug to reduce the occlusion effect. Classical lumped element models usually simplify the earplug as an impedance connected to the earcanal cavity. Intricate couplings of the earplug with the earcanal wall and the earcanal cavity are thus neglected. Finite element models can account for these couplings and thus make it possible to study the earplug contribution. In this work, the physical mechanisms that explain the earplug contribution to the objective occlusion effect are investigated. For this purpose, a 2-D axi-symmetric finite element model of an outer ear is used. Two types of earplug (foam and silicone) as well as two insertion depths (medium and deep) are considered. The earplug influence is interpreted in terms of volume velocity imposed to the earcanal cavity and related to its mechanical properties and its insertion depth.

## *Contributed Paper*

12:20

**5aNSb4. Are fitness instructors at risk of hearing impairment?** Mariola Sliwinska-Kowalska (Audiol. and Phoniatrics Clinic, Nofer Inst. of Occupational Medicine, 8 Sw. Teresy Str., Lodz 91-348, Poland, mariola.sliwinska@imp.lodz.pl), Kamil R. Zaborowski (Dept. of Physical Hazards, Nofer Inst. of Occupational Medicine, Lodz, Poland), Anna Wolniakowska (Audiol. and Phoniatrics Clinic, Nofer Inst. of Occupational Medicine, Lodz, Poland), Adam Dudarewicz, and Malgorzata Pawlaczyk-Luszczynska (Dept. of Physical Hazards, Nofer Inst. of Occupational Medicine, Lodz, Poland)

The entertainment industry can pose a risk of hearing impairment to employees. The aim of the study is to assess whether occupational exposure to loud sounds (music) is related to temporary hearing threshold shifts in fitness instructors. The study comprised a total of 29 fitness instructors working in fitness clubs. The noise dosimeters were used to determine individual noise exposure during typical 1-2 hours exercises conducting by the instructors. Pure-tone audiometry and Hearing in Noise Test (HINT) were performed before and just after the session to assess temporary changes in hearing. The A-weighted equivalent-continuous sound pressure level ($L_{Aeq,T}$) during typical exercises session ranged from 76.3 to 96 dB. Referring to the 3-dB equal energy rule, such exposures may exceed the upper exposure action value (85 dB) set for the occupational settings by the Directive 2003/10/EC. A significant temporary threshold shift (at least 6 dB) was observed in 33 audiograms (40 %). No impairment of speech understanding was observed in the HINT. Fitness instructors may constitute a population at increased risk of hearing impairment. Further studies are needed to conclude on hearing protection programs in this group of workers.

# Session 5aPAa

## Physical Acoustics, Engineering Acoustics, Structural Acoustics and Vibration, and Biomedical Acoustics: Acoustofluidics V

Charles Thompson, Cochair
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Kedar Chitale, Cochair
*Flodesign Sonics, Inc., 380 Main Street, Wilbraham, MA 01095*

Max Denis, Cochair
*University of the District of Columbia, 4200 Connecticut Ave. NW, Washington, D.C. 20008*

**Chair's Introduction—9:30**

### *Contributed Papers*

**9:35**

**5aPAa1. Acoustic trapping of sub-wavelength microparticles and cells in resonant cylindrical shells.** Qin Lin (Inst. of Biomedical and Health Eng., Shenzhen Institutes of Adv. Technol., Chinese Acad. of Sci., 1068 Xueyuan Ave., Shenzhen University Town, Shenzhen, Guangdong 518055, China, qin.lin@siat.ac.cn), Feiyan Cai, Fei Li, Xiangxiang Xia, and Hairong Zheng (Inst. of Biomedical and Health Eng., Shenzhen Institutes of Adv. Technol.,Chinese Acad. of Sci., Shenzhen, China)

Acoustic tweezers based on the sound beam systems hold the promise of contactless manipulation of microparticles. However, in these conventional diffraction-limited system, acoustic diffraction severely limits the trapping strength and the minimum size of the trapped particles. Here, we theoretically propose and experimentally demonstrates that a simple cylindrical shell based acoustic tweezers can be utilized for trapping of sub-wavelength particles and cells with a radius as small as 1/400 of the corresponding acoustic wavelength. This mechanism is attributed to the significantly enhanced acoustic radiation force originating from the resonant excitation of low order circumferential mode intrinsically existing in the cylindrical shell, which is a highly localized field around its surfaces and breaking the diffraction limit. We further demonstrate that the manipulation ability of these tweezers is significantly stronger than that of traditional standing wave based acoustic tweezers, which can significantly reduce physiological damage to cells or other biological objects arising from the thermal effects. Thus, the cylindrical shells based acoustic tweezers are simple, disposable, low cost, biocompatible, and functional, with applications including 3-D bio-printing, cell culturing and tissue engineering.

**9:55**

**5aPAa2. Flexible transportation of microparticles and cells in phononic crystal based acoustofluidic channel.** Feiyan Cai (Shenzhen Institutes of Adv. Technol.,Chinese Acad. of Sci., 1068 Xueyuan Ave., Shenzhen University Town, Shenzhen 518055, China, fy.cai@siat.ac.cn), Fei Li (Shenzhen Institutes of Adv. Technol., Chinese Acad. of Sci., Shenzhen, China), Zhengyou Liu (Key Lab. of Artificial Micro- and Nano-Structures of Ministry of Education and School of Phys. and Technol., Wuhan Univ., Wuhan, China), Likun Zhang (Univ. of MS, Oxford, MS), and Hairong Zheng (Shenzhen Institutes of Adv. Technol.,Chinese Acad. of Sci., Shenzhen, China)

As one of the prominent technologies for non-contact manipulations, acoustic tweezer devices have used both standing waves and sound beams to trap particles at a fixed point, i.e., static positioning. However, dynamic manipulations of microparticles, cells, and living organisms are demanded in various micro/nanotechnologies, such as biomedical sensors, imaging devices, diagnostic tools, etc. The dynamic manipulation of microparticles in real-time requires time-variant acoustic fields. Here, a novel way for three-dimensional dynamic manipulation of microparticles and cells in the acoustofluidic channel is proposed by using a phononic crystal plate that modulates the acoustic field in a desired manner. This approach is superior to the conventional dynamic manipulations based on tuning phases of standing waves or moving sound sources. The phononic crystal based manipulations benefit from the resonant excitation of specific modes in the phononic crystal plate. The switch between different resonant modes in the phononic crystal plate enables the flexible switching between trapping and transportation, sets up a basis for completely flexible manipulation of microparticles in acoustofluidic channel.

10:15

**5aPAa3. An acousto-gravitational balance in climbing films of water and oil for separating liquid mixtures.** Amihai Horesh (Chemical Eng., Technion - Israel Inst. of Technol., La Jolla, CA), Anna Zigelman, Daniel Khaikin, Mackenzi Karnilaw (Chemical Eng., Technion - Israel Inst. of Technol., Haifa, Israel), and Ofer Manor (Chemical Eng., Technion - Israel Inst. of Technol., Technion, Wolfson Faculty of Chemical Eng., Haifa, Israel, manoro@technion.ac.il)

We present the climb of water menisci and silicon oil films up a vertical acoustic actuator. The experimental system is comprised of a vertical 20 MHz-frequency surface acoustic wave (SAW) device whose edge is submerged in a reservoir of liquid. Upon the application of power, the SAW propagates down the device toward the liquid reservoir. We monitor the response of the liquid to the SAW. Partially wetting water and surfactant solutions naturally satisfy a finite three phase contact angle between the liquid, vapor, and the solid substrate of the device, which gives rise to curved liquid menisci in the absence of SAW excitation. In the presence of the SAW, the menisci rise up the device to obtain steady state heights above the level of liquid in the reservoir. In contrast, silicon oil supports a vanishing contact angle with the SAW device, which gives rise to flat films of oil. In the presence of the SAW, silicon oil films are found to continuously climb up the device at a constant velocity, which is similar to the corresponding velocity on a horizontal SAW device. We use the difference in the response of oil and water to separate liquid mixtures.

10:35

**5aPAa4. Scaleable production of microbubbles using an ultrasound-modulated microfluidic device.** Walid Messaoudi, Dario Carugo (Univ. of Southampton, Southampton, United Kingdom), and Eleanor P. Stride (Univ. of Oxford, Inst. of Biomedical Eng., Oxford OX3 7DQ, United Kingdom, eleanor.stride@eng.ox.ac.uk)

Surfactant-coated gas microbubbles are widely used as contrast agents in ultrasound imaging and increasingly for therapeutic applications. It has been shown that the acoustic response of microbubbles is determined by their size and coating properties and hence depends upon both their chemical composition and the manufacturing technique used to produce them. We have previously presented a hybrid device consisting of a simple microfluidic T-junction with an integrated ultrasound transducer that provides superior production rates and microbubble stability compared with conventional microfluidic systems but with significantly better microbubble uniformity than standard emulsification techniques. The maximum production rate was, however, still limited compared to industrial methods. In the present study a new device was developed that enables production of $>10^8$ microbubbles per second using a single device with a mean bubble diameter of 1.4 $\mu$m without degrading microbubble uniformity. Production rates of $>10^9$ microbubbles per second can be achieved through parallel operation of multiple channels within a single device; comparable with bulk emulsification but without the risk of contamination and/or degradation of sensitive components.

5a FRI. AM

# Session 5aPAb

## Physical Acoustics, Engineering Acoustics, Structural Acoustics and Vibration, and Biomedical Acoustics: Acoustofluidics VI

Charles Thompson, Cochair
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**Chair's Introduction—11:15**

### *Invited Paper*

**11:20**

**5aPAb1. Three-dimensional trapping and assembly with synchronized spherical acoustical vortices.** Zhixiong Gong (Univ. Lille, CNRS, Centrale Lille, Yncréa ISEN, Univ. Polytechnique Hauts-de-France, UMR 8520 - IEMN, F- 59000, IEMN, Cité Scientifique Ave. Henri Poincaré CS 60069, Lille 59652, France, zhixiong.gong@iemn.fr) and Michael Baudoin (Univ. Lille, CNRS, Centrale Lille, Yncréa ISEN, Univ. Polytechnique Hauts-de-France, UMR 8520 - IEMN, F- 59000, Lille, France)

Micro-objects and micro-organisms trapping and assembly with acoustical tweezers would open new horizons in microrobotics and microbiology, e.g., selective cells fusion and aggregation. Our previous work [Gong and Baudoin, *Phys. Rev. Appl.* **12**, 024045 (2019)] demonstrates theoretically in *two dimensions (2-D)* the possibility to trap and assemble small particles compared to the wavelength with synchronized acoustical tweezers based on cylindrical acoustical vortices. However, there is no trap in the beam's propagation direction since the cylindrical acoustical vortex is progressive along their central axis, leading to the fact that particles are only pushed or pulled (not trapped) in this direction and hence are mainly limited to 2-D operations. In this work, we extend our previous analysis and show theoretically that particles can be trapped and assembled in three dimensions with synchronized spherical vortices. We show that the particles can be approached both laterally (similar to the 2-D synchronized cylindrical vortices) and axially (extra to the 2-D case) and we determine the maximum assembly speed by balancing the critical radiation force and the Stokes' drag force. These theoretical results provide guidelines to design selective acoustical tweezers able to trap and assemble small particles in three dimensions.

### *Contributed Papers*

**11:40**

**5aPAb2. Acoustic radiation force acting on a heavy particle in a standing wave can be dominated by the acoustic microstreaming.** Alen Pavlic (ETH Zurich, Inst. for Mech. Systems, Tannenstrasse 3, CLA H25, Zurich 8092, Switzerland, apavlic@ethz.ch), Thierry Baasch, and Jürg Dual (ETH Zurich, Inst. for Mech. Systems, Zurich, Switzerland)

When an acoustic wave scatters on a particle the acoustic radiation force and the microstreaming appear as non-linear time-averaged effects. Although they appear simultaneously, the microstreaming, which is driven by viscous losses, is often neglected in the theoretical modeling of the acoustic radiation force. Here, we investigate the contribution of the acoustic microstreaming to the acoustic radiation force acting on a small elastic spherical particle placed into an ultrasonic standing wave [T. Baasch, A. Pavlic, and J. Dual, *Phys. Rev. E* **100**(6), 061102 (2019)]. The compressible Navier-Stokes equations are solved up to second-order in terms of the small Mach number using a finite element method. Our study shows that above a certain viscosity, when the viscous boundary layer thickness to particle radius ratio is sufficiently large, the contribution of the microstreaming dominates the acoustic radiation force and defines the stable position of the particle, provided that the particle is sufficiently dense. In such cases (e.g., combination of a copper particle of 1 $\mu$m radius in a mineral oil), our theory predicts migration of the particle to the pressure antinode.

12:00

**5aPAb3. Highly efficient acoustophoretic single cell-supernatant separation inside nanoliter droplets.** Michael Gerlt (ETH Zurich, Tannenstrasse 3, Zurich 8092, Switzerland, gerlt@imes.mavt.ethz.ch), Dominik Haidas, Alexandre Ratschat, Philipp Suter, Petra Dittrich, and Jürg Dual (ETH Zurich, Zurich, Switzerland)

This contribution reports a novel method for cell-supernatant separation inside nanoliter droplets with separation efficiencies for single yeast cells of 100% at flow speeds of 2 mm/s. Cells are focused in nanoliter droplets using bulk acoustic waves before the droplet is split at a bifurcation into a supernatant- and a cell-containing droplet. The chip design is optimized using criteria found through experiments and numerical simulations. The separation efficiency is measured in detail as a function of droplet speed, size, split ratio, and particle concentration for PS beads. Subsequently, yeast cells, mammalian cells and bacteria were introduced into the system to test its versatility. The method enables upconcentration of cells, cell washing, and the separation of supernatant in droplet microfluidic applications.

### *Invited Paper*

12:20

**5aPAb4. Equivalence of three-dimensional acoustic radiation force and torque formulas based on angular spectrum and multipole expansion methods.** Zhixiong Gong (Univ. Lille, CNRS, Centrale Lille, Yncréa ISEN, Univ. Polytechnique Hauts-de-France, UMR 8520 - IEMN, F- 59000, IEMN, Cité Scientifique Ave. Henri Poincaré CS 60069, Lille 59652, France, zhixiong.gong@iemn.fr) and Michael Baudoin (Univ. Lille, CNRS, Centrale Lille, Yncréa ISEN, Univ. Polytechnique Hauts-de-France, UMR 8520 - IEMN, F- 59000, Lille, France)

Acoustic radiation force (ARF) or torque (ART) can be derived by integrating the time-averaged linear and angular radiation stress tensor (the flux of momentum tensor) over a far-field standard spherical surface including an arbitrarily shaped target. Following this idea, there are three sets of independent original derivation on the ARF ([Silva, *J. Acoust. Soc. Am.* **130**, 3541–3544 (2011)] and [Baresh *et al.*, *J. Acoust. Soc. Am.* **133**, 25–36 (2013)] based on the multipole expansion method, and [Sapozhnikov and Bailey, *J. Acoust. Soc. Am.* **133**, 661–676 (2013)] based on the angular spectrum method); and two independent work on the ART ([Silva *et al.*, *EPL* **97**, 54003 (2012)] based on the multipole expansion method and [Gong and Bandoin, *J. Acoust. Soc. Am.* (under review; 2020)] based on the angular spectrum method). In this work, we formally establish the equivalence between the expressions obtained by these two methods with the relation of beam shape coefficients at different elementary waves of the field for both the force and torque. The differences between the existing formulas in the literature are indicated and explained. The respective advantages of each form are discussed.

FRIDAY MORNING, 11 DECEMBER 2020 — 9:30 A.M. TO 11:00 A.M.

## Session 5aPPa

## Psychological and Physiological Acoustics: New Approaches and Virtual Reality I

Gregory M. Ellis, Chair
*Communication Sciences and Disorders, Northwestern University, 2240 Campus Drive, Chicago, IL 60626*

**Chair's Introduction—9:30**

### *Contributed Papers*

9:35

**5aPPa1. Estimation of the category of notch frequency bins of the individual head-related transfer functions using the anthropometry of the listener's pinnae.** Kazuhiro Iida (Adv. Media, Chiba Inst. of Technol., 2-17-1, Tsudanuma, Narashino, Chiba 2750016, Japan, kazuhiro.iida@it-chiba.ac.jp), Orie Nishiyama, and Tsubasa Aizaki (Graduate School, Chiba Inst. of Technol., Narashino, Japan)

An authentic approach to present three-dimensional acoustical sensation to everyone is to provide head-related transfer functions (HRTFs) that are exactly adapted to each listener. This approach is called the individualization of HRTFs. In order to generate individualized HRTFs, the present study proposes a modeling method of HRTFs that extracts the frequency bins in which spectral notches (N1 and N2) are included. The notch frequency bins for N1 and N2 were classified into two categories based on the just noticeable difference in notch frequency with regard to the vertical angle perception of a sound image. Then, discriminant analyses were carried out as objective variables of the category of each of the N1 and N2 frequency bins and as explanatory variables of ten anthropometric parameters of pinnae. The results show that the categories of N1 and N2 frequency bins can be estimated with accuracies of 78.9% and 77.8%, respectively.

9:55

**5aPPa2. Semi-supervised self-adjustment fine-tuning procedure for hearing aids for asymmetrical hearing loss.** Jonathan A. Gößwein (Hearing-, Speech- and Audio Technol., Fraunhofer IDMT, Marie-Curie-Str. 2, Oldenburg, Lower Saxony 26129, Germany, jonathan.goesswein@idmt.fraunhofer.de), Rainer Huber, Tobias Bruns (Hearing-, Speech- and Audio Technol., Fraunhofer IDMT, Oldenburg, Lower Saxony, Germany), Josef Chalupper (European Res. Ctr., Adv. Bionics, Hannover, Lower Saxony, Germany), Martin Kinkel (Res. and Development, KIND Hearing Aids, Grossburgwedel, Lower Saxony, Germany), Jan Rennies (Hearing-, Speech- and Audio Technol., Fraunhofer IDMT, Oldenburg, Lower Saxony, Germany), and Birger Kollmeier (Medical Phys., Carl von Ossietzky Univ. Oldenburg, Oldenburg, Lower Saxony, Germany)

The individual fitting of hearing aids is still a challenge and usually requires several sessions. The audiologist typically fine-tunes the hearing aids based on the patient's reported perception. Recent research investigated the alternative of empowering the patient by means of self adjustment. However, all known studies on self-adjustment procedures have so far focused on symmetric hearing loss and a symmetrical signal modification adjustable by the user. It is therefore still unknown how to deal with severe asymmetric hearing losses. In this study, we examined a previously evaluated self-adjustment procedure for symmetric hearing losses with respect to its applicability for asymmetric hearing losses. For this purpose, experienced hearing-aid users with asymmetric hearing loss were fitted with real hearing aids and equipped with a self-adjustment user interface. Each fitting was performed in several realistic sound scenes in two conditions: first, the two hearing aids were fitted separately; second, both hearing aids were fitted in a coupled way and then fine-tuned separately. In addition to the comparison between the gain settings resulting from the self-adjustments the study examined also subjective sound impressions such as the balance of the sound in both ears.

10:15

**5aPPa3. Linearity, orthogonality, and resolution of psychoacoustic sonification for multidimensional data.** Tim Ziemer (Bremen Spatial Cognition Ctr., Univ. of Bremen, Enrique-Schmidt-Str. 5, Bremen 28359, Germany, ziemer@uni-bremen.de) and Holger Schultheis (Bremen Spatial Cognition Ctr., Univ. of Bremen, Bremen, Germany)

Sonification is the use of sound to communicate data to a listener. In sonification designs that lack psychoacoustic considerations, dimensions tends to be perceived as nonlinear, interfering, and as having a low resolution. As a solution to these issues, we developed a "psychoacoustic sonification," in which single dimensions or variables of multidimensional or multivariate data are mapped to independent psychoacoustic quantities instead of orthogonal physical quantities. Distances along different direction are mapped to the speed of cyclicchroma change, the beat frequency, the degree of roughness, brightness and fullness. In this paper we present results of 3experiments that evaluate the linearity, orthogonality, and resolution of the dimensions. In experiment 1, participants identify a sonified target field in multiple two-dimensional spaces. Results indicate the orthogonality of the dimensions. In experiment 2, the sonification guides the participants through a two-dimensional space to find an invisible target. The results verify the linearity and orthogonality of the dimensions and indicate a high resolution. Experiment 3 is a JND experiment according to the transformed up-down method, which demonstrated the high resolution of the dimensions. We conclude that applying psychoacoustic knowledge in the signal processing of sonification is a powerful tool for multidimensional data sonification.

10:35

**5aPPa4. Sound source localization with various ambisonics orders in virtual reality.** Thirsa Huisman (Tech. Univ. of Denmark, Ørsteds Plads, Bygning 352, Kgs. Lyngby 2800, Denmark, thuis@dtu.dk), Axel Ahrens (Tech. Univ. of Denmark, Kgs. Lyngby, Denmark), and Ewen MacDonald (Univ. of Waterloo, Waterloo, ON, Canada)

To reproduce realistic audio-visual scenarios in the laboratory, Ambisonics is often used to present a sound field over loudspeakers and virtual reality glasses are used to present visual information. However, the application of both technologies, Ambisonics and virtual reality glasses, might affect the spatial cues for auditory localization, resulting in reduction of the localization accuracy. Furthermore, the combination of both technologies might introduce further errors. Here, we investigated how a head-mounted display affects the localization of virtual sound sources produced using either 1st, 3rd, 5th or 11th order Ambisonics with and without visual information. Preliminary results suggest that there is an effect of Ambisonics order on localization accuracy, mostly in the first order Ambisonics, but that the virtual reality glasses do not add an additional error. The localization error with first order ambisonics remained even with visual information as the perceived location matched better with other visual sources (i.e., nearby loudspeakers). Thus, virtual reality can be used in combination with Ambisonics without additional localization errors, but for localization accuracy it is better to use higher order Ambisonics.

# Session 5aPPb

## Psychological and Physiological Acoustics: New Approaches and Virtual Reality II

Gregory M. Ellis, Chair

*Communication Sciences and Disorders, Northwestern University, 2240 Campus Drive, Chicago, IL 60626*

**Chair's Introduction—11:15**

### *Contributed Papers*

**11:20**

**5aPPb1. The freedom to move around—Introduction of the Sonova real life lab.** Stefan Klockgether (R&D, Sonova AG, Laubisrütistrasse 28, Stäfa, Zürich 8712, Switzerland, stefan.klockgether@sonova.com), Diego Ulloa Sanchez, Charlotte Vercammen, and R. Peter Derleth (R&D, Sonova AG, Stäfa, Zürich, Switzerland)

In experiments, control is everything. If parameters, procedures or the environment cannot be controlled thoroughly, any experiment lacks repeatability and hardly any findings can be generalized. In psychoacoustics and audiology research, important aspects of perception in real life are sacrificed in favor of control. The new "Real Life Lab" at Sonova allows audiological research to take a big step forward towards real life environments by bringing the freedom to move around in acoustic environments to controlled laboratory conditions. The lab provides a large stage where persons can move around freely, interact with acoustic or real objects, or with simulated or real people. The stage is surrounded by a framework of loudspeakers to present acoustic content from all directions and screens to present visual content in the horizontal plain. Any motion by persons on the stage can be tracked in real time with a motion capturing system. The motion capturing data can be passively tracked during any experiment or it can be actively used to alter or trigger audio and video reproduction. This contribution shows the technical capabilities of the lab together with some insights on the first experiment on the motion behavior in an orientation task in a complex acoustic environment.

**11:40**

**5aPPb2. Effects of hearing aid processing on speech intelligibility in virtual restaurant settings.** Gregory M. Ellis (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Chicago, IL 60626, gregory.ellis@northwestern.edu) and Pamela E. Souza (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

Hearing-impaired listeners struggle to understand speech in complex environments like restaurants. Environmental factors (signal-to-noise ratio and reverberation time) and hearing aid processing (digital noise reduction and wide dynamic range compression [WDRC]) have been shown to affect speech intelligibility in these contexts. We examine the independent and synergistic effects of these environmental factors and processing strategies on speech intelligibility. Virtual space techniques were used to build a virtual restaurant. The target talker was located directly in front of the listener and competing talkers were located to the listener's right in various positions around the restaurant. Signal-to-noise ratio was varied randomly, centered on 0 dB with a standard deviation of 2 dB. Reverberation time was either 0.8 or 1.8 s. Hearing aid processing was simulated using custom software. WDRC was either fast or slow with the following attack/release times: 5/50 ms and 100/2000 ms. Digital noise reduction was either on or off. Customized gain and frequency shaping were applied to the signals. Listeners heard low-context sentences through insert earphones and repeated the target sentence. Responses were scored based on the number of correctly repeated keywords. Preliminary results show that the effect of WDRC depended on reverberation time. [Work supported by NIH.]

**12:00**

**5aPPb3. Production and use of multimedia speech content for perceptual experiments in virtual environments.** Philip W. Robinson (Facebook Reality Labs, 9845 Willows Rd., Redmond, WA 00076, philrob22@gmail.com), Lindsey Kishline, and Scott Colburn (Facebook Reality Labs, Redmond, WA)

Consumer virtual reality devices have become inexpensive and readily available, and offer high quality motion tracking, low latency, and sufficient resolution to conduct ecologically valid perceptual experiments. Unfortunately, high quality multimedia source material can be difficult to produce or obtain due to the specialized equipment and facilities required for capture. To facilitate a wide array of perceptual experiments in multi-modal speech perception, we have generated a multimedia corpus that includes stereoscopic and spherical videos, as well as anechoic audio. These materials can easily be placed into an interactive virtual reality environment and delivered over a head-mounted display to evaluate ventriloquism, lip-reading, spatial release from masking, or other perceptual effects that depend on audio-visual integration. The corpus replicates the Coordinate Response Measure sentences, as well as the Harvard IEEE corpus word list. Subjects were recorded at three different distances to preserve accurate binocular disparities in the videos and can be positioned at arbitrary azimuthal positions. The corpus and a simple virtual reality application for positioning and viewing the videos on a head-mounted display have been made publicly available online for download. We will detail the methods used to produce this content, as well as use of the accompanying viewing application.

**12:20**

**5aPPb4. One step at a time to feel lighter: Understanding the impact of sound and smell on body image perception.** Giada Brianza (Univ. of Sussex, Falmer, Brighton BN19RH, United Kingdom, g.brianza@sussex.ac.uk) and Gianluca Memoli (Univ. of Sussex, Brighton, United Kingdom)

How people mentally represent their body image does not always match their actual body. A negative body image perception (i.e., BIP) can cause risks of eating disorders, isolation, and emotional disease. Thus, being able to manipulate this perception through technology can open up the opportunity to increase healthy behaviours. Previous works showed that technology can be used to change people's BIP combining visual and tactile stimulation. However, can other senses make the difference? Can audio and smell impact on our BIP? We based our work on a well-established link between walking sounds and the perceived walker's weight: heavier bodies produce lower spectral mode sounds than lighter bodies. We ran a multisensory user-study in which we altered in real-time the frequency spectra of the sound of participants' footstep. In the meantime, we delivered previously selected scents. Our results show that the combination of audio and scent stimuli can be used to make participants feel lighter or heavier and that highly arousing scents (e.g., lemon) enhance the effect of sound. We discuss limitations (e.g., the use of headphones and wired devices) and potentials of our findings (e.g., the use of multisensory to overcome BIP misperception). With this work, we want to make the community aware of the power of audio on BIP, and inform future research towards the creation of novel virtual multisensory experiences and devices that can positively impact the way we feel about ourselves.

## Session 5aSAa

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

Benjamin C. Treweek, Cochair
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Trevor Jerome, Cochair
*Naval Surface Warfare Center, Carderock Division, West Bethesda, MD*

**Chair's Introduction—9:30**

### *Contributed Papers*

**9:35**

**5aSAa1. Vibro-acoustic analysis and optimization of hearing aids.** Prachee Priyadarshinee (Mech. Eng., National Univ. of Singapore, #21-256B, 38 College Ave. East, Singapore 138601, Singapore, a0135595@u.nus.edu)

This work presents the vibro-acoustic analysis and optimization of behind-the-ear(BTE) hearing aids. The aim was to improve the overall performance and reduce the feedback caused by the vibrations transmitted due to the tranducer. The vibro-acoustic model also improved the vibration isolation mounts to reduce the vibrations transmitted through the hearing aid, while keeping the packaging compact and without sacrificing the performance accuracy.

**9:55**

**5aSAa2. An improved procedure for shape preserving design of vibrating structures using topology optimization.** Olavo M. Silva (Multidisciplinary Optimization Group, Federal Univ. of Santa Catarina, Campus Universitario Trindade, Centro Tecnologico, Florianópolis 88040900, Brazil, olavo@lva.ufsc.br)

The shape preserving design approach by topology optimization was firstly introduced for linear static problems, in which the local strain energy of a controlled domain inside a given structure is constrained with a minor upper bound beyond zero, designated to suppress its warping deformation (and, consequently, preserving its shape even when the whole body deforms). Shape preserving design applied to structures subjected to harmonic loads was explored in a recent article, but without reporting the difficulties found throughout the numerical processes. In that research, the local dynamic compliance was used in its classical form as a local measure of deformation/warping. In the present work, the author shows that instabilities arising from the use of such measure as a constraint function make the procedure unstable and slow. The frequency behavior of the classical dynamic compliance is directly related to the involved displacement and force responses, presenting antiresonances within a given frequency range. This fact may cause instabilities and early convergence of the optimization process. Thus, it is suggested the constraint on the local time-averaged potential energy for a better controlling of deformation/warping of specific regions in a given structure. Some examples are presented and discussed, which indicate the effectiveness of the proposed procedure. In addition, more physical meaning can be found by using this measure instead of the classical dynamic compliance as a constraint function.

**10:15**

**5aSAa3. Neumann accelerations of generalized polynomial chaos expansions for larger dynamic systems.** Allison Kaminski (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, allison9@bu.edu), James McDaniel, and Alyssa Liem (Mech. Eng., Boston Univ., Boston, MA)

This work proposes a method for accelerating generalized polynomial chaos (gPC) expansions, using Neumann series, for uncertainty quantification of complex structures with uncertain damping and spring parameters. Often, classical Monte Carlo simulations are used for uncertainty quantification, however, for larger systems and frequency sweeps this method becomes very computationally expensive. Neumann series have been used with Monte Carlo simulations to decrease computation times, however, the Neumann series does not always converge. More recently gPC expansions have been used for uncertainty quantification. gPC is a stochastic spectral method that develops a surrogate model, whose coefficients can be used to calculate statistical moments. gPC requires sampling the response at predetermined quadrature points, which may become expensive for large systems. This study proposes a new approach where Neumann series are used to calculate the response at gPC quadrature points, in order to accelerate gPC computation times. The validity of this method is shown using large, viscously damped systems. Damping and stiffening elements fully connect degrees of freedom to each other and to ground. First the damping elements are taken as the uncertain parameter, then stiffening elements are taken as the uncertain parameter, and frequency dependent responses such as power dissipated are calculated.

**10:35**

**5aSAa4. Power flow analysis for quantifying structural modifications.** Jon Young (Appl. Res. Lab, 340 Toftrees Ave. Apt. 258, State College, PA 16803, jyoung.engr@gmail.com) and Kyle Myers (Appl. Res. Lab, Bellefonte, PA)

Structural modification is the process of changing the physical properties of a structure to achieve a desired objective. For structural-acoustic problems, the objective is usually to minimize structure-borne and air-borne sound, which are generally quantified in terms of power. Mathematically, additions made to the mass, damping, and stiffness matrices, which form the structure's impedance matrix, will change the amount of power that can be input to, dissipated by, flow through, and radiated by the structure. After making structural modifications, the impedance matrix can be expressed as the sum of the impedances of the original structure and the modifications made to the structure. An analytical means of determining how structural modifications affect the aforementioned power metrics with respect to the original structure is examined by utilizing the properties of the inverse of sums of matrices, with emphasis on using spring, dashpot, point mass, and generalized impedances as the structural modifications. Several examples are developed in order to validate the method for numerically generated data, and parametrically show how the modifications affect the power metrics in physical and modal space.

# Session 5aSAb

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

Benjamin C. Treweek, Cochair
*Computational Solid Mechanics & Structural Dynamics, Sandia National Laboratories, P.O. Box 5800, Albuquerque, NM 87185*

Trevor Jerome, Cochair
*Naval Surface Warfare Center, Carderock Division, West Bethesda, MD*

**Chair's Introduction—11:15**

### *Contributed Papers*

**11:20**

**5aSAb1. The role of resonance in the extreme value statistics of flow-induced response.** Connor J. McCluskey (Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, cxm1198@psu.edu), Manton J. Guers, and Stephen C. Conlon (Penn State Univ., State College, PA)

Extreme value statics (EVS) are used to predict outlying loads that greatly damage structures. EVS are commonly applied to random environmental loads, such as sea states. However, the output response from structure's transfer function has different statistics from the input load. Accurate prediction of maximum loads require analysis of the random response, which includes considering worst-case conditions such as resonance. This work aims to investigate the EVS of the flow-induced response from the applied load and the resonance of the first bending mode. In a water tunnel, an upstream cylinder was used to shed a wake onto a cantilever fin. The tunnel flow speed was increased to allow harmonics of the cylinder's vortex shedding frequency to excite the fin's first bending mode. The measured response was band-pass filtered to separate stiffness-controlled and resonance frequency bands. Extreme values of each record were modeled with the Generalized Extreme Value distribution. The experimental results are investigated to provide insight on how resonance impacts EVS and the consideration of resonance in accounting for extreme responses.

**11:40**

**5aSAb2. Mechanical resonators as rheological sensors for oil and gas process monitoring.** Miguel Gonzalez (Aramco Res. Center—Houston, Aramco Americas, 17155 Park Row, Houston, TX 77084, miguel.gonzalez@aramcoamericas.com)

Oil and gas operations deal with a wide variety of non-Newtonian fluids that are either being extracted or are utilized for specific tasks and it is critical to monitor their rheological properties. Here, we present different rheological measurement platforms based on mechanical resonators for use in drilling muds, enhanced-oil-recovery (EOR) polymer solutions, and hydraulic fracturing fluids. The characteristic resonator sizes used in these platforms span from the nanoscale to the macroscale, and therefore also span a broad range of time-scales for probing the high-frequency viscoelastic response of the fluids. First, an electromechanical tuning fork resonator for in-tank viscosity/density measurements of drilling fluids is described. The resonator works as the frequency-defining element of an oscillator circuit. When the feedback is disconnected, the resonance frequency and oscillation decay time are used to obtain the viscosity and density of the fluid consistently and in real-time. Second, I describe the use of miniature resonating devices, from the nanometer to the millimeter scale, to obtain the viscoelastic response of polymer fluids used in EOR and hydraulic fracturing operations. Finally, I discuss the possibility of using the nonlinear vibrational response from the damping in a non-Newtonian fluid to infer shear rheology-dependent properties of the fluid.

**12:00**

**5aSAb3. Windowing technique in determining the importance of sound propagation through vehicle glazing systems.** Pranab Saha (Kolano and Saha Engineers, Inc., 3559 Sashabaw Rd., Waterford, MI 48329, prsaha@kandse.com), Satyajeet Deshpande (Kolano and Saha Engineers, Inc., Lincoln, NE), William Fisher, and Vikram Bhatia (Corning, Inc., Corning, NY)

The glazing system of a vehicle plays a very significant role in developing an efficient sound package system. However, in developing the sound package treatments in the vehicle, although one studies the importance of various paths through which sound is travelling, often the glazing system is overlooked. This paper discusses the windowing technique methodology that is used for identifying the importance of paths and the setup that was used to determine the importance of individual glazing units and the collective glazing system in the overall sound propagation through the body panels of a SUV. Results show that a significant amount of sound energy propagates from outside to inside the vehicle through all glazing units combined than through the rest of the vehicle.

## Session 5aSCa

## Speech Communication: Developing a Cross-Platform Federated Code Repository for Speech Research I

Charles H. Redmon, Cochair
*Linguistics, University of Kansas, 427 Blake Hall, University of Kansas, Lawrence, KS 66045*

Matthew C. Kelley, Cochair
*Univ. of Alberta, Edmonton, AB, Canada*

Benjamin V. Tucker, Cochair
*Linguistics, University of Alberta, 4-32 Assiniboia Hall, University of Alberta, Edmonton, T6G 2E7, Canada*

**Chair's Introduction—9:30**

### *Invited Papers*

**9:35**

**5aSCa1. Developing a cross-platform federated code repository for speech research.** Charles H. Redmon (Linguist, Univ. of Kansas, 933 Rhode Island St., Apt. 5, Lawrence, KS 66044, redmon@ku.edu), Matthew C. Kelley, and Benjamin V. Tucker (Linguist, Univ. of AB, Edmonton, AB, Canada)

The increasing proliferation of code, scripts, and programming language libraries for use in speech science research raises concerns over the maintenance and organization of this disparate code base. These concerns have motivated us to explore the development a federated code repository under the umbrella of the Acoustical Society of America. Our primary goal for this repository is to serve as a central point from which researchers in the speech sciences can access code and data from the community that adheres to a set of established standards for documentation. The code will also be reviewed by other researchers for errors, vulnerabilities, and algorithmic misspecifications. Further, this repository will serve as a starting point for the future development of parallel libraries in open-source languages like R, Python, and Julia. In this talk, we will outline our plans for the repository and identify key open challenges and questions to be answered by contributors regarding repository structure, the code submission and review process, and documentation standards to adopt.

**9:55**

**5aSCa2. Developing and maintaining the *Phonological CorpusTools* software.** Kathleen C. Hall (Linguist, Univ. of BC, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, kathleen.hall@ubc.ca), Roger Y. Lo, and Stanley Nam (Linguist, Univ. of BC, Vancouver, BC, Canada)

*Phonological CorpusTools* (http://phonologicalcorpustools.github.io/CorpusTools/) is a free, open-source, cross-platform software written in Python 3 that comes with a graphical user interface. It is designed to facilitate the analysis of phonological patterns in transcribed data, calculating characteristics such as phonotactic probability, functional load, neighbourhood density, informativity, and degree of complementary distribution. Having such a tool helps increase the reproducibility of quantitative phonological corpus analysis. We discuss some of the challenges we have encountered in developing this software (and its companion, *Sign Language Phonetic Annotator & Analyzer*(SLP-AA)), including (1) the difficulty of providing long-term, ongoing development and support in a world of changing technologies and (2) the difficulty of making a software tool that is easily accessible to users with a wide variety of starting data types. Having centralized, staffed, and funded repositories (cf. Alveo: http://alveo.edu.au/) would help mitigate some of the practical difficulties of sharing, distributing, and maintaining such resources. We also suggest that when developing specific tools, it is important to invest considerable time and resources in the input/output interface, which in turn facilitates both the uploading of data by disparate users and the workflow of transferring data from one application to another.

**10:15**

**5aSCa3. Using praat for high-quality speech manipulation and illustration: Recommended practices and demonstrations.** Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Shevlin Hall Rm 115, Minneapolis, MN 55455, mwinn@umn.edu)

Praat is widely used and freely available software that is designed with speech acoustics in mind. Despite its popularity, there is little standardization on the development of features and scripts, causing duplication of work across labs. This presentation has two goals: (1) highlighting a number of features that could be useful to kickstart such standardization, and (2) demonstration of scripts that could be useful for experimenters looking to create high-quality perceptual stimuli or classroom demonstrations. Recommendations will be given for variable assignment, documentation, increasing stimulus naturalness, and incorporating auditory scaling rather than linear

interpolation for both frequency and intensity. Demonstrations will include manipulations of vowel formant contours, voice onset time, pitch contour, and fricative spectra, all using natural speech. Additionally, resources will be highlighted that allow users to easily compare outputs of sound modifications like spectral filtering. Examples will be given of data visualization within Praat to verify clean manipulations, as well as creating publication-quality images using R while handling objects from Praat.

**10:35**

**5aSCa4. Building bridges between software ecosystems: Parselmouth, a Python interface for Praat.** Yannick Jadoul (Vrije Universiteit Brussel, Pleinlaan 2, Elsene 1050, Belgium, Yannick.Jadoul@ai.vub.ac.be)

The value of software depends on more than just the intrinsic quality of the algorithms it implements. In order to be useful, existing software needs to be combinable with functionality in other software libraries. In particular, this holds for research software: modularity and the ability to be integrated in prominent existing software ecosystems are key to correct and easy reuse. Over the last few years, I have created Parselmouth, a Python library for Praat, a software package widely used in speech science. Compared to previous projects offering a thin interface to Praat, Parselmouth's goal is to provide a full-fledged Python library that integrates efficiently into the larger Python ecosystem. In this talk, I will briefly demonstrate Parselmouth and its functionality, then discuss the motivation to create Parselmouth, what different goals had to be considered, and why I believe the current solution strikes a desirable tradeoff between these conflicting goals. Moreover, after working out a technical solution, another important aspect is documenting, distributing, and publicizing the new software package, especially for the wildly varying user base of Praat. While Parselmouth still has a long way to go here, I will also give an overview of how I have approached these aspects.

FRIDAY MORNING, 11 DECEMBER 2020 — 11:15 A.M. TO 12:35 P.M.

## Session 5aSCb

## Speech Communication: Developing a Cross-Platform Federated Code Repository for Speech Research II

Charles H. Redmon, Cochair
*Linguistics, University of Kansas, 427 Blake Hall, University of Kansas, Lawrence, KS 66045*

Matthew C. Kelley, Cochair
*Univ. of Alberta, Edmonton, AB, Canada*

Benjamin V. Tucker, Cochair
*Linguistics, University of Alberta, 4-32 Assiniboia Hall, University of Alberta, Edmonton, T6G 2E7, Canada*

**Chair's Introduction—11:15**

### *Invited Paper*

**11:20**

**5aSCb1. A cloud-computing platform for developing and evaluating vocal biomarkers based on home audio recordings: Resources for large-scale data processing and analysis.** Gordon Ramsay (Dept. of Pediatrics, Emory Univ., Marcus Autism Ctr., 1920 Briarcliff Rd. NE, Atlanta, GA 30329, gordon.ramsay@emory.edu)

This presentation describes the design and construction of a comprehensive cloud-computing platform for implementing and evaluating vocal biomarkers, both for assessing typical development of speech and language and detecting deviations in autism. As part of a prospective longitudinal study of early vocal development (NIH P50 MH100029), whole-day home audio recordings were collected from more than 300 children every month from birth to three years using LENA, providing long-term, large-scale sampling of each child's natural environment. The database contains more than 40 000 h of raw audio data, post-processed acoustic measures, event timing statistics, hand-coded and automated annotations, as well as sociodemographic and clinical metadata, the storage, integration and processing of which has presented significant new challenges. Flexible cloud solutions have been adopted to allow secure cross-platform shared access and referencing of the data. Customized modular "C" code libraries for physical modeling, spectral analysis, temporal event labeling, voice activity detection, speaker diarization and developmental profiling have been developed, based on standard BLAS/LAPACK libraries optimized for device-independent virtual machines, and integrated with an interactive graphical interface for signal analysis and labeling. Resources and specific challenges for large-scale analysis of infant vocalizations will be discussed.

## *Contributed Paper*

11:40

**5aSCb2. A novel tool for automated assessment of listener transcripts in speech intelligibility studies.** Hans Rutger Bosker (Max Planck Inst. for Psycholinguistics, P.O. Box 310, Nijmegen 6500 AH, The Netherlands, HansRutger.Bosker@mpi.nl)

In the field of speech perception, many studies assess the intelligibility of spoken stimuli by means of verbal repetition ("repeat back what you hear") or transcription tasks ("type out what you hear"). The intelligibility of a given stimulus is then often expressed in terms of percentage of words correctly reported from the target stimulus. Yet scoring the participants' raw responses for words correctly identified from the target stimulus is a time-consuming task, and hence resource-intensive. Moreover, there is no consensus on what protocol to use for the human scoring, limiting the reliability of human scores. The present paper evaluates various forms of "fuzzy string matching" between participants' responses and target sentences as automated metrics of listener transcript accuracy. Fuzzy string matching is identified as a consistent, efficient, and accurate method for automated assessment of listener transcripts, as evidenced by high correlations with human-generated scores ($r$ = 0.922) and a strong relationship to acoustic markers of speech intelligibility. Thus, fuzzy string matching provides a practical tool for assessment of listener transcript accuracy in large-scale speech intelligibility studies.

FRIDAY MORNING, 11 DECEMBER 2020 — 9:30 A.M. TO 11:00 A.M.

# Session 5aSPa

# Signal Processing in Acoustics: General Topics in Signal Processing in Acoustics IV

James C. Preisig, Chair
*Jpanalytics, LLC, 82 Technology Park Drive, East Falmouth, MA 02536*

**Chair's Introduction—9:30**

## *Contributed Papers*

9:35

**5aSPa1. Persistent scatterers detection from phase statistics of multi-view synthetic aperture sonar imaging.** Angeliki Xenaki (Ctr. for Maritime Res. and Experimentation, Viale S. Bartolomeo, La Spezia 19126, Italy, angeliki.xenaki@cmre.nato.int), Yan Pailhas, and Roberto Sabatini (Ctr. for Maritime Res. and Experimentation, La Spezia, Italy)

Persistent scatterers (PS) refer to stationary point reflectors that exhibit strong phase stability, irrespective of their amplitude. Synthetic aperture sonar (SAS) systems provide high-resolution seafloor imaging by adding coherently the backscattered waves from multiple views along the synthetic antenna. Traditionally, PS detection relies on the magnitude of the resulting image, thus it is susceptible to high levels of speckle noise. This study demonstrates that PS can be detected more reliably from pixel-wise phase statistics of multi-view complex SAS images. Phase-based feature extraction aims to facilitate automatic target recognition tasks without training data.

9:55

**5aSPa2. Development of sonar simulation program for synthetic aperture processing with arbitrary sensor motion.** Sea-Moon Kim (Korea Res. Inst. of Ships and Ocean Eng., 32 Yuseong-daero 1312 Beon-gil, Yuseong-gu, Daejeon 34103, South Korea, smkim@kriso.re.kr) and Sung-Hoon Byun (Korea Res. Inst. of Ships and Ocean Eng., Daejeon, South Korea)

Due to high image resolution capability, the synthetic aperture sonar (SAS) is widely used for search and identification of underwater objects on seafloor. However, the image resolution of SAS is greatly influenced by sensor array movement and it is very important to estimate precise sensor positions. In order to verify SAS signal processing performance when the motion is out of the planned route, a simulation software based on MATLAB script has been developed which can simulate any arbitrary motion of transmitters and receivers. In addition, the motion and attitude of targets can be described by any arbitrary function of time for future study on identification of moving targets. To describe as close to an actual SAS system, the directivity patterns of transmitters and receivers can be applied with any arbitrary shape. It is also possible to have any reflectivity pattern of targets which is a function of incident and scattered angles with respect to the axis fixed on the targets. In this presentation SAS processing results with various situations as well as verification of the software will be presented. [This work is financially supported by the research project PES3570 funded by KRISO.]

10:15

**5aSPa3. Using predicted target state to inform transmit waveform selection in active sonar.** Venakata Veeramachaneni (Elec. and Comput. Eng., George Mason Univ., Abu Dhabi, United Arab Emirates) and Jill K. Nelson (Elec. and Comput. Eng., George Mason Univ., 4400 University Dr., MSN 1G5, Fairfax, VA 22030, jnelson@gmu.edu)

Continuous-wave (CW) and frequency-modulated (FM) transmit pulses are commonly used in active sonar systems. CW pulses allow for measurement of target Doppler but have relatively poor range accuracy. In contrast, FM pulses are insensitive to Doppler but allow for improved range accuracy relative to CW. Active sonars often operate in clutter-dominated environments, where choosing a CW pulse for a high Doppler target provides better tracking capability, while an FM pulse is more effective for a low Doppler target. A challenge, however, is that the target Doppler not known before transmission. At present, the complementary strengths of CW and FM pulses can be exploited only by an operator who actively chooses between them, but requiring an operator to select the transmitted waveform on a ping-by-ping basis presents an unreasonable burden. In this talk, we describe the Predicted State-Based Selection (PSBS) algorithm, which uses an estimate of the target Doppler, derived from a state prediction produced by the tracking filter, to select the transmitted pulse. Monte Carlo simulations are conducted to evaluate the performance and behavior of the PSBS algorithm. Results show that PSBS improves target localization estimates by 7.7% on average relative to transmission of the same waveform for all pings.

10:35

**5aSPa4. The cutting parameters dependent vibration monitoring method for machine tools.** Mingxin Hui (Inst. of Acoust., Chinese Acad. of Sci.No. 21, West Rd. of North 4th Ring, Haidian District, Beijing 100190, China, huimingxin18@mails.ucas.ac.cn), Jing Wang, Bin Liu, Xun Wang, Xiaobin Cheng, and Jun Yang (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

The on-line monitoring of the machine tools attracts growing interest for the operation of the smart factories. The quality of the workpieces and the productivity of the manufacturing can be maintained through the monitoring and optimization. The vibration signal reflects irregular characteristics of the fault conditions, such as the tool wear and chatter. The corresponding features can be extracted and recognized using the conventional signal processing method in the time and frequency domain. However, the vibration features are process-dependent. Due to the dynamic changes of cutting parameters, the variance of the cutting forces and the consequent frequency response between different cutting parameters could be larger than that between the regular conditions and fault conditions in the identical cutting parameters. This paper combined the vibration signal with the real-time cutting parameters and toolpath calculated by the numerical control codes to analysis the cutting condition of the machining process. A sub-band segmentation method based on spindle speed is proposed. The accuracy and applicability are also discussed.

FRIDAY MORNING, 11 DECEMBER 2020 — 11:15 A.M. TO 12:45 P.M.

## Session 5aSPb

## Signal Processing in Acoustics: General Topics in Signal Processing in Acoustics V

Vaibhav Chavali, Cochair
*Electrical and Computer Engineering, George Mason University, 4400 University Dr., MS 1G5, Fairfax, VA 22030*

David J. Geroski, Cochair
*Applied Physics, University of Michigan, 2313 Packard St., Apt. A103, Ann Arbor, MI 48104*

**Chair's Introduction—11:15**

### *Contributed Papers*

11:20

**5aSPb1. A sub-band filter design approach for sound field reproduction.** Yongjie Zhuang (Mech. Eng., Purdue Univ., Ray W. Herrick Labs., 177 S Russell St., West Lafayette, IN 47907, zhuang32@purdue.edu), Guochenhao Song, and Yangfan Liu (Mech. Eng., Purdue Univ., West Lafayette, IN)

The purpose of sound field reproduction is to use loudspeakers to produce desired sound at particular locations in a given environment, which has a wide range of applications such as virtual reality, etc. The computational load required to design and implement filters involved in sound reproduction systems can be significant, especially when the desired sound has rich information over a wide frequency band. To reduce the computational load, sub-band filtering approaches are usually used in sound reproduction systems. In the present work, an approach is proposed to design the sub-band filters used in sound reproduction systems in a more convenient way, where the filter design problem is formulated into a convex optimization problem. A detailed analysis has been conducted on how to specify the response characteristics of each sub-band and how different sub-band filters can be combined into one full band filter in the design and implementation of the system. Results also show that even if the sub-band filter structure is not necessary, this approach can also be applied to reduce the computational load in designing inverse filters when the plant responses involve relatively large differences in delay time among different frequency bands.

**11:40**

**5aSPb2. Ambisonics and blind source separation in virtual acoustics: Sound field reproduction of separated sources.** Louis J. Dermagne (Université de Sherbrooke, Groupe d'Acoustique de l'Université de Sherbrooke, 1945, rue du Montagnais, Sherbrooke, QC J1K2Z3, Canada, louis.dermagne@usherbrooke.ca), Philippe-Aubert Gauthier (Université du Québec à Montréal., École des Arts Visuels et Médiatiques, Montréal, QC, Canada), and Alain Berry (Université de Sherbrooke, Groupe d'Acoustique de l'Université de Sherbrooke, Sherbrooke, QC, Canada)

Blind source separation (BSS) has many applications: sound scene analysis, speech recognition, medical signal processing, etc. However, most of these applications concern the temporal separation of signals. Studies have shown the effectiveness of separation in the ambisonic domain with spherical microphone recordings. Thanks to the ambisonic approach, it is possible to separate the directions of arrival of the sources. As such, BSS becomes a promising tool for sound field reproduction with loudspeakers arrays (Wave Field Sythesis or Higher-Order Ambisonics). Thanks to spherical microphone arrays and Ambisonics principle, both spatial and temporal information are available. Therefore, it would be possible to reproduce the individual sound field of each separated source. Thus, one can remove a given source from a recording and reproduce the remaining sound field. The main objective of this work is to reproduce the sound field of one of the captured sources by removing the rest of it (sources or noise). The first part of the paper presents the methods for the BSS and corresponding sound field reproduction. The second part presents simulation results and investigates effect of measurement noise, spatial source separation, and reflection. [Work supported by NSERC Discovery grant.]

**12:00**

**5aSPb3. Obtaining far-field spherical directivities from arbitrarily shaped arrays using the Helmholtz equation least-squares method.** Samuel Bellows (Dept. of Phys., Brigham Young Univ., Provo, UT 84602, samuel.bellows11@gmail.com) and Timothy W. Leishman (Brigham Young Univ., Provo, UT)

Commonly, directivities of sound sources are measured using surrounding spherical arrays. However, in some cases, such as when measurement hardware is limited, or when an array shape would ideally conform to the radiating object's geometry, obtaining spherical far-field directivities using an arbitrarily shaped array could be useful. This work illustrates how the Helmholtz equation least-squares method applies to accomplish this task. Both numerical models and the measured directivities of various guitar amplifiers validate the technique.

**12:20**

**5aSPb4. Acoustic source centering of musical instrument directivities using acoustical holography.** Samuel Bellows (Dept. of Phys., Brigham Young Univ., Provo, UT 84602, samuel.bellows11@gmail.com) and Timothy W. Leishman (Brigham Young Univ., Provo, UT)

The acoustic center of a source is commonly defined as the point from which spherical wavefronts appear to diverge. Measuring directivities of sound sources with a surrounding spherical array whose geometric origin is not aligned with the acoustic center of the source can lead to distortions in directivity patterns. Thus, a method is desired to obtain both the far-field directivity pattern and to determine the acoustic center of a source. This work illustrates how acoustical holography can identify the reference frame from which spherical waves diverge by studying various musical instruments' acoustic centers.

FRIDAY MORNING, 11 DECEMBER 2020 9:30 A.M. TO 11:00 A.M.

## Session 5aUWa

## Underwater Acoustics: Underwater Sound Transmission I

Brian T. Hefner, Chair

*Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105*

**Chair's Introduction—9:30**

### *Contributed Papers*

**9:35**

**5aUWa1. Sound propagation in Arctic-like waveguides.** Konstantin Dmitriev, Alexey Lipavskiy, Ivan Pankov (Phys. Faculty, Acoust. Dept., Moscow State Univ., Moscow, Russian Federation), Sergei Sergeev (Phys. Faculty, Acoust. Dept., Moscow State Univ., Leninskie Gory, Moscow 119991, Russian Federation, sergeev@aesc.msu.ru), and Grigorii Fomin (Phys. Faculty, Acoust. Dept., Moscow State Univ., Moscow, Russian Federation)

Sound propagation in Arctic-like waveguides (regions with a strong interaction of the acoustic field propagating in the water layer with both ice cover and the bottom) is complex phenomenon. The authors have performed more than two dozen experiments in different seasons on the shelf of the White Sea and in Moscow region both in the active regime and in the noise interferometry regime. In the active regime, it was demonstrated that the presence of ice leads to a decrease in the critical frequencies of the modes. If the bottom gives off gas, its accumulation under the ice layer significantly affects the boundary conditions. Flexural waves propagate over the ice and can radiate energy into the water layer. Interference occurs and is significant, especially near and below the cutoff frequency of the waveguide. Ice movement causes crackling. Using the acoustic noise caused by the crackling as a signal (noise interferometry regime) on the Arctic shelf allowed to solve the inverse problem and to reconstruct the sea parameters, for example, tidal activity. Using several horizontally spaced hydrophones, it was possible to study the processes of the propagation of individual cracks in the ice.

9:55

**5aUWa2. Assessment of statistical features of the diffused acoustic field in a test section of a cavitation tunnel.** Romuald Boucheron (DGA Hydrodynamics, Chaussée du Vexin, Val de Reuil 27105, France, romuald.boucheron@intradef.gouv.fr)

Noise measurements in a cavitation tunnel is nowadays the preferred approach for predicting the underwater noise radiated by a propeller at sea. The accuracy of the prediction is linked to the similitude laws used to transform the model scale measurements to the full-scale ones. Besides, this accuracy is obviously linked to the accuracy of the model scale measurements. The knowledge of the acoustical features of the facility used to perform such measurements is then one of the key point of such measurements. The proposed communication presents the recent results obtained in the test section of the Large Cavitation Tunnel (GTH at Val-de-Reuil) to describe the statistics of the acoustic diffused field. The results shows that the diffused field comply spatially very well with the Gumbel distribution for narrow band levels (expressed in dB for the power spectral density). The limit of the statistical domain (defined as the "Schroeder frequency") is also estimated thanks to the experimental measurements performed inside the test section.

10:15

**5aUWa3. Influence of the warping time parameters on dedopplerization efficiency in the case of a rotating source in cavitation tunnel.** Romuald Boucheron (DGA Hydrodynamics, Chaussée du Vexin, Val de Reuil 27105, France, romuald.boucheron@intradef.gouv.fr)

Cavitation tunnels are hydrodynamic facilities used mainly to observe cavitation and to analyze the performances of maritime devices likes propellers or foils. These experimental approaches, consisting in a model scale measurements extrapolated to full scale with similitude laws, are nowadays the preferred methods to predict the underwater noise radiated by a vessel at sea. The characterization of the different acoustic sources (localization and strength mainly) is one of the key point to better understand the whole mechanism that generates the total underwater noise. The particular case of a rotating source imposes a Doppler effect that disturbs the frequency content of the signal measured by a fixed hydrophone. The communication presents the results obtained by simulation of the measured signal generated by a rotating source. A warping time function is defined with geometrical parameters and used to perform a dedopplerization of the measured signal. The influence of each of those parameters is presented focusing on the performance of the expected dedopplerization. Consequently, different strategies are proposed to define an adapted criterion representing the accuracy of the dedopplerization effect.

10:35

**5aUWa4. Modeling environmental uncertainty in ocean propagation simulations.** Jay R. Johnson (Dept. of Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109, jayrj@umich.edu), Brandon M. Lee, and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

The ocean is a dynamic random environment and predictive ocean acoustics tasks generally require knowledge of environmental parameters. To understand the extent to which environmental uncertainty impacts acoustic field calculations, each of the environmental parameters can be considered a random variable and to contain temporal and/or range dependence. Many computational ocean acoustics tasks, such as assessing the impact of anthropogenic noise sources, designing ocean propagation experiments, and finding commercial aircraft emergency locator beacons, would benefit from leveraging the underlying uncertainty in these properties to yield more accurate and robust predictions. However, appropriate sampling of ocean environment properties to quantify acoustic uncertainties can be difficult. While in-situ studies can rely on repeated experimental data collection and direct measurements of uncertainties to deduce the impact of varying environmental state, computational studies do not often have efficient or robust tools for incorporating uncertainty. In this talk, a framework is presented to sample uncertain ocean environmental states using open-source databases to predict acoustic uncertainties. Example Monte-Carlo simulation results for acoustic amplitude and phase at frequencies from 100 to 500 Hz are presented for propagation ranges up to 100 km.

# Session 5aUWb

## Underwater Acoustics: Underwater Sound Transmission II

Brian T. Hefner, Chair
*Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105*

**Chair's Introduction—11:15**

### *Contributed Papers*

**11:20**

**5aUWb1. Modeling and measurement of relative intensity of acoustic arrivals within the beaufort duct.** Jessica Desrochers (Ocean Eng., The Univ. of Rhode Island, 13 Gilroy St. Apt. 2, Newport, RI 02840, jfothergill@my.uri.edu), Lora Van Uffelen (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Sarah E. Webster (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Matthew A. Dzieciuch, and Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA)

The Beaufort Duct, located at approximately 175 m depth in the Canada Basin, allows for long-range acoustic propagation from sources moored within the duct. As part of the Canada Basin Acoustic Propagation/Glider Experiment (CANAPE/CABAGE), LFM sweeps with frequencies around 250 Hz were transmitted by broadband moored sources and received by two Seagliders traversing between them. In addition to receiving acoustic data, these Seagliders measured temperature, pressure, and salinity, providing measurements of the sound speed profile throughout the CANAPE study region. These profiles were used as inputs for broadband acoustic parabolic equation predictions. Measured data receptions within the duct show intense peak arrivals prior to the final cutoff that are higher in relative intensity than predicted. The duct properties of the measured sound speed profiles are parameterized and acoustic predictions using actual profiles as well as idealized profiles are used to study the predictability of these intense peaks within the duct.

**11:40**

**5aUWb2. Low frequency sound absorption in the Arctic Ocean: Update on the impact of ocean acidification.** David Browning (Browning Biotech, 139 Old North Rd., Kingston, RI 02881, decibeldb@aol.com) and Peter Herstein (Browning Biotech, Kingston, RI)

The principal mechanism for low frequency sound absorption in seawater is a boron relaxation reaction that is pH dependent: the lower the pH; the lower the absorption. In 1987, Mellen *et al.* [*J. Acoust. Soc. Am.* **82**, S30 (1987)] extensively computed the low frequency absorption in the Arctic Ocean. Since that time the carbon dioxide (CO2) level in the atmosphere has been continually increasing, and in 2014 Browning and Herstein reported the corresponding change in ocean pH and resulting change in seawater sound absorption that had occurred [*J. Acoust. Soc. Am.* **135**, 2306 (2014)]. This paper now provides a further update to the present time on the changing Arctic Ocean low frequency sound absorption and indicates what the result might beif the United Nations' goal of reaching equilibrium in 12 years is met.

**12:00**

**5aUWb3. The update of Urick's *Principles of Underwater Sound.*** Charles H. Wiseman (Peninsula Publishing, 1630 Post Rd. East, Unit 312, Westport, CT 06880, chaswiseman30@gmail.com)

For over 50 years *Principles of Underwater Sound* by Robert Urick has been a widely used book on underwater acoustics and sonar for practicing engineers, scientists, technicians, project managers, teachers and students and it is in the last stages of an extensive update to be released this spring. The book encapsulates the fundamentals and phenomena of underwater sound as applied to the Sonar Equation, the heart of prediction of sonar performance, and the book's contents lie squarely in the middle between theory at one end and practical technology at the other. The editor and contributors of the update have attempted to imbue the new edition with the same "blue collar" spirit of Urick's original work but with so much new technology to address since the present edition was published in 1983, it's been difficult to keep the updated printed edition to a convenient size. Therefore, in addition to the traditional printed book edition, we are introducing a new *e-book* edition which will provide more detail than the *printed* edition. This presentation includes: (1) Descriptions of the two new formats (2) Examples of updated technology, and (3) New subjects: sonar transmissions' effect on the health of marine mammals; sea floor mapping; locating downed airliners; and search plans.

**12:20**

**5aUWb4. Ray-based methods in PC SWAT.** Denton Woods (NSWC PCD, 110 Vernon Ave., Panama City, FL 32407, denton.woods@navy.mil)

The Personal Computer Shallow Water Acoustic Toolset (PC SWAT), developed at the Naval Surface Warfare Center Panama City Division (NSWC PCD), is a ray-based simulator capable of running multiple types of sonar simulation scenarios. Recent PC SWAT capability enhancements, such as arbitrarily shaped targets, target occlusion by sea bottom features, and parallelization efforts, will be discussed. These improvements foster greater physical accuracy and fidelity for target scattering simulations. [Work supported by the Office of Naval Research.]

# Session 5pAAa

## Architectural Acoustics: Architectural Acoustics Potpourri IV

Jonas Braasch, Chair
*School of Architecture, Rensselaer Polytechnic Institute, School of Architecture, 110 8th Street, Troy, NY 12180*

**Chair's Introduction—1:05**

### *Contributed Papers*

**1:10**

**5pAAa1. The role of architectural acoustics in advancing the aural tradition during the Safavid era.** Nima Farzaneh (Architecture, Rensselaer Polytechnic Institue, 1520 6th Ave., Apt. 407, Troy, NY 12180, farzan@rpi.edu) and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

The Safavid era was a significant ruling dynasty in Iran (1501–1736). This period enabled a rich collection of art and architecture as well as scientific and engineering innovations. Naqsh-e-Jahan Square, the apotheosis of the Safavids' urban design, hosts four monumental structures on each side of the plaza. On the south side, the Imam Mosque is an exemplary building in which the Iranian scholar Shaykh Baha'i implemented his knowledge of the science of sound and hyperbolic mathematics to solve acoustical problems. The result is a tranquil soundscape that allows for synchronized prayer rituals with intelligible speeches for up to 15 000 people. The Ali Qapu palace, with its magnificent music room, the Sheikh Lotfollah Mosque, the royal family spiritual practice space, and the bazaar portal connecting the great plaza to the heart of the city, each contribute differently to the sonic identity of this complex. This research focuses on the mechanism of sound energy exchange, energy decay pattern due to acoustic coupling, volumetric design, and the impact of material selection and ornamentation in creating the sonic landscape. In a broader scheme, the goal is to find the connection between the created soundscape and the intended function of the spaces.

**1:30**

**5pAAa2. Clay pots in Ottoman architecture: A discussion on their acoustical employment.** Gülnihan Atay (Dept. of Architecture, Bilkent Univ., Ankara 06800, Turkey, gulnihan.atay@bilkent.edu.tr) and Zühre Sü Gül (Dept. of Architecture, Bilkent Univ., Ankara, Turkey)

The employment of the clay pottery as an acoustical element first appeared in Vitruvius's texts in the antiquity. Up to this date, their precise contribution to the acoustic quality of the space has remained vague despite many studies examining their employment as cavity resonators in medieval liturgical architecture throughout Europe and Near East. They are also observed in the substantial number of structures that belong to the Classical Ottoman Architecture, some of which were designed by Sinan, the architect laureate of Ottoman Empire. One of his works, Süleymaniye Mosque, which contains substantial number of pots in its grand dome exhibited significantly high reverberation times especially in the low frequency ranges in the recent acoustic measurements. Such long decays set forth the question whether the clay pots were designed to act as Helmholtz resonators that would control the low frequencies in their original state. Thus, this study aims to exhibit the state of the art of the clay potteries utilized as cavity resonators in the Ottoman Architecture and set a ground to compare their utilization with their contemporaries around the world as well as initiating further studies on the working principles of the cavity resonators in Ottoman Architecture.

**1:50**

**5pAAa3. Sound field studies in archaeological fieldwork: near or far, free or direct or diffuse field—How does it matter?** Steven J. Waller (Rock Art Acoust., 1952 Sonoma Ln., Lemon Grove, CA 91945, wallersj@yahoo.com)

The relatively new discipline of archaeoacoustics faces challenges regarding the wide variety of prehistoric and historic environments to be studied. This paper focuses on the need to consider different types of acoustic techniques to use in various situations. Hypothesis testing can potentially lead to different results depending on the type of acoustic testing performed. Nearfield versus farfield approaches can yield disparate datasets, so the methodology chosen can have significant impact on the conclusions. How do sound reflecting surfaces affect the sound power radiated from a source, or received by a listener, and influence its apparent directionality and cultural perception? Examples of acoustic research results will be presented, including studies of deep cave paintings of Europe, shallow painted shelters of India and Australia, and petroglyphs carved into the walls of canyons and cliff faces in North America.

**2:10**

**5pAAa4. The acoustics of the "clapping circle" at Purdue University.** Elspeth A. Wing (Interdisciplinary Eng., School of Eng. Education, Purdue Univ., Neil Armstrong Hall of Eng. Rm. 1300, 701 W. Stadium Ave., West Lafayette, IN 47907, winge@purdue.edu), Steven J. Herr (Elec. Eng., Purdue Univ., West Lafayette, IN), Alexander D. Petty (Interdisciplinary Eng., Purdue Univ., West Lafayette, IN), Alexander K. Dufour (Elec. Eng., Purdue Univ., West Lafayette, IN), Frederick H. Hoham (Interdisciplinary Eng., Purdue Univ., West Lafayette, IN), Morgan E. Merrill (Elec. Eng., Purdue Univ., West Lafayette, IN), Donovan W. Samphier (Audio Eng. Technol., Purdue Univ., West Lafayette, IN), Weimin Thor (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., West Lafayette, IN), Kushagra Singh (Mech. Eng., Purdue Univ., West Lafayette, IN), Yutong Xue (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., West Lafayette, IN), Davin H. Huston (Audio Eng. Technol., Purdue Univ., West Lafayette, IN), and J. S. Bolton (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., West Lafayette, IN)

The unusual acoustical properties of a particular landscape architecture feature of Academy Park on the Purdue University campus have been the subject of speculation for years. The feature, known informally as the "clapping circle," consists of sixty-six concentric rings of stone tiles. When someone claps while standing at the middle of the circle, they hear a high-pitched squeak immediately afterwards. Experiments were conducted by the Purdue student chapters of the Acoustical Society of America and the Audio Engineering Society to characterize this effect. The response to a clap played from an omnidirectional speaker placed at the center of the circle was recorded using a microphone positioned above the loudspeaker. Spectrograms of the recorded responses revealed the squeak to consist of a descending tone at around 1500 Hz, and its harmonics. This tone disappeared from the spectrogram when the tile rings were covered with absorbing

blankets. A mathematical model based on scattering from the gaps between the tile rings reproduced the descending frequency of the squeak, and reproduced the effect of the source and receiver height on the rate of change of frequency. Thus, it was concluded that the squeak is an example of repetition pitch produced by the tile formation.

FRIDAY AFTERNOON, 11 DECEMBER 2020 2:50 P.M. TO 4:20 P.M.

# Session 5pAAb

## Architectural Acoustics: Architectural Acoustics Potpourri V

Timothy Hsu, Chair
*Music and Arts Technology, Indiana University - Purdue University, Indianapolis, 535 W. Michigan St., IT 371, Indianapolis, IN 46202*

**Chair's Introduction—2:50**

### *Invited Paper*

**2:55**

**5pAAb1. Non-cuboid iterative room optimizer.** Peter D'Antonio (Res., REDI Acoust., LLC, 99 South St., Passaic, NJ 07055, pdantonio@rpgacoustic.com), Renaldi Petrolli, and John Storyk (Res., REDI Acoust., LLC, Highland, NY)

In past years, various iterative optimization programs emerged to separately determine optimal room ratios, sources and listening positions in perfectly reflective cuboid rooms, using the image source model. This approach fails to account for scattering, phase change at the boundary and cannot be extended to non-cuboid rooms. The present work presents a solution, using the boundary element method (BEM) to compute the frequency response at low frequencies, considering the effects of the complex admittances of the boundaries and all acoustical elements inside the room. With BEM as its engine, a genetic algorithm was developed to optimize source and receiver positions simultaneously with the room geometry, within architectural restraints. Given real-world limitations, optimizing source and listener positioning and room geometry is not always an option. Therefore, a damper module was added, which addresses low frequency acoustic treatment. By using different transfer matrix models, the acoustical behavior of different low-frequency pressure absorbers can be modeled and inserted into the BEM simulation, to evaluate the change in the room's acoustic field and frequency response at the receiving positions of interest. 3-D waterfalls are used to illustrate the modal decay following optimization. Examples will be presented.

### *Contributed Papers*

**3:15**

**5pAAb2. Simulation-validated generative design of acoustical arrays with diffusion and absorption.** Timothy Hsu (Music and Arts Technol., Indiana Univ. - Purdue Univ., Indianapolis, 535 W. Michigan St., IT 371, Indianapolis, IN 46202, hsut@iu.edu) and Jonathan Dessi-Olive (Dept. of Architecture, Kansas State Univ., Manhattan, KS)

This paper presents several new results for on-going efforts in combining computational design methods and acoustical needs. Past investigations employed rule-based design strategies that expanded upon current monolithic deployment of quadratic residue diffusers (QRD) in the field. The algorithm generated novel, and visually diverse arrays of N = 7 QRD panels. The arrays generated visually strayed from the typical appearance of QRD arrays; appearing more expressive as rules were applied. Furthermore, evaluation metrics including a visual complexity coefficient and 3-D polar plots from PML-FEM simulations provided visual evidence that a relationship exists between the appearance of an array and its corresponding acoustical response. Building on this previous work, representative arrays generated by the output of the algorithms with expanded parameters will be presented alongside updated evaluation metrics. The algorithm uses an expanded set of simulation-validated rules that include non-square grid configurations, multiple design frequencies, a varying number of wells, and incorporate zones of absorption. These new arrays, along with results and discussion of acoustical response to these rules, will be presented. Using these expanded parameters absorption, field of room acoustics design is opened up to wide-ranging visual design possibilities that are fully cooperative and incorporated with the acoustic response.

**3:35**

**5pAAb3. Design of efficient low-frequency sound absorbers using an array of Helmholtz Resonators.** Vidhya Rajendran (Dept. of Architecture, Univ. of Washington, 3950 University Way NE, Seattle, WA 98105, vidhya.rj28@gmail.com), Tomás I. Méndez Echenagucia (Dept. of Architecture, Univ. of Washington, Seattle, WA), and Andrew A. Piacsek (Phys., Central Washington Univ., Ellensburg, WA)

This paper describes the design process of a low-frequency sound absorptive panel composed of differently tuned Helmholtz Resonators. Given the size and fabrication constraints relevant for applications in the building sector, we focus on cylindrical and spiral resonators with embedded necks that are space-efficient and can achieve perfect absorption. Apart from the design of an individual resonator, the design process involves

modifying the mutual interaction between the resonators to improve the absorption performance of the whole panel. Additional resistance in the necks is considered as a means to shift the absorption towards lower frequencies. Panel sizes of 0.2 m × 0.2 m loaded with 9 or 16 resonators were tested in an impedance tube built specifically for measuring low-frequency absorption. The measured absorption performance of these panels is consistent with the theoretical predictions. Of particular interest is the impact of the arrangement of the resonators in the panel on its absorption performance, which has been overlooked in the theoretical models. As predicted, the experiments show a shift in absorption performance towards low-frequencies with the inclusion of neck embedding and additional resistance in the neck, and without changing the geometry of the panel.

**3:55**

**5pAAb4. Analysis and acoustical modeling of transit noise reflections to residential locations.** Alex Maurer (Intertek, 60 Charlesgate E, Boston, MA, MA 02215, alexander.maurer@intertek.com)

As development progresses in urban areas, it is common that new projects are developed adjacent to existing public transportation systems. Varying in dimensions and façade materials, these new developments introduce new surfaces off of which sound can reflect to locations previously not impacted. The acoustical ramifications of the facades of new developments are often ignored and can create reflections to residential locations that are additive to the direct sound of the transit system. These reflections can be subjectively regarded by nearby residents as bothersome or intrusive. We have studied one such case of reflected transit noise adversely impacting nearby residents. We measured short duration sound pressure levels ($L_p$s) of the transit activity near the affected residential locations and created a three-dimensional sound propagation model to better understand the impact of transit noise reflections off of two nearby residential developments, comparing the results of our study with Federal Transit Authority Guidelines.

FRIDAY AFTERNOON, 11 DECEMBER 2020 — 1:05 P.M. TO 2:35 P.M.

## Session 5pBAa

## Biomedical Acoustics: General Biomedical Acoustics: Therapeutics III

John M. Cormack, Chair
*Department of Medicine, University of Pittsburgh, Pittsburgh, PA 15261*

**Chair's Introduction—1:05**

### *Contributed Papers*

**1:10**

**5pBAa1. Effect of sonication parameters on the efficacy of focused ultrasound and microbubble-mediated blood-spinal cord barrier opening using short-burst, phase keying exposures.** Stecia-Marie P. Fletcher (Sunnybrook Research Inst., 2075 Bayview Ave., M7-302, Toronto, ON M4N 3M5, Canada, sfletcher@sri.utoronto.ca), Min Choi, Ranjith Ramesh (Sunnybrook Reserch Inst., Toronto, ON, Canada), and Meaghan O'Reilly (Sunnybrook Res. Inst., Toronto, ON, Canada)

Focused ultrasound (FUS) and microbubbles can open the blood-spinal cord barrier (BSCB) and enhance therapeutic delivery. Short-burst phase keying (SBPK) exposures (pulse-train of closely timed short bursts) have been developed to address clinical-scale, spine specific targeting challenges in a dual-aperture configuration, and have led to successful BSCB opening (BSCBO) in animal models. Here we study the effect of varying sonication parameters *in vivo*. The effect of varying acoustic pressure (P, n = 4), burst length (BL, n = 3), burst repetition frequency (BRF, n = 4), pulse-train length (PL, n = 4), and total treatment duration ($\tau$, n = 4), compared with a control sonication (P = 0.28 MPa, BL = 2 cycles, BRF = 20 kHz, PL = 10 ms, $\tau$ = 120 s, frequency = 514 kHz, PRF = 1 Hz), was investigated in Sprague Dawley rats (3-4 locations/spinal cord). BSCBO was assessed using T1-weighted contrast-enhanced MRI. Statistical significance was assessed using a paired t-test ($p < 0.05$). Increased P led to increased MRI enhancement (0.23 MPa: 17.8 ± 4.9%, 0.28 MPa: 25.9 ± 8.0%, 0.33 MPa: 33.9 ± 8.8%). $\tau$ = 300 s showed increased enhancement compared with the control (29.5 ± 8.7 vs 22.2 ± 2.8%, p = 0.03), while BL = 5 showed a trend towards increasing enhancement, although without significance (36.6 ± 7.2% vs. 31.2 ± 3.0%, p = 0.17). Varying PL or BRF did not impact mean enhancement. Preliminary results show that increasing treatment duration improves the efficacy of SBPK FUS-induced BSCBO and increasing burst length may have some benefit. Future work will include histological assessment of tissue damage.

**1:30**

**5pBAa2. A novel model using *Lumbricus terrestris*, earthworms, for microvessel rupture induced by high-intensity therapeutic ultrasound exposure in the presence of microbubbles.** Asis Lopez (U.S. Food and Drug Administration, 6823 St Charles Ave., New Orleans, LA 70118, asis.lopez@gmail.com), Yaswitha Mikkilineni (Virginia Commonwealth Univ., Richmond, VA), Shayna Berman (Montgomery College, Takoma Park, MD), Damir Khismatullin (Biomedical Eng., Tulane, New Orleans, LA), Gregory T. Clement (U.S. Food and Drug Administration, Cleveland, OH), and Matthew R. Myers (U.S. Food and Drug Administration, Silver Spring, MD)

We present the earthworm as a useful model for studying vasculature rupture induced by High-Intensity Therapeutic Ultrasound (HITU) with Microbubbles. Although vertebrates are indispensable to biomedical

research, studies are often limited by factors such as cost, lengthy internal review, and ethical considerations. An extensive database of vessel-rupture probabilities and times was created for both HITU and HITU + Microbubbles as a function of critical parameters, including microbubble dosage and size, and ultrasound operating frequency and intensity. The earthworm model allows the large number of trials to be performed that enable identification of the critical characteristics of bubbles, blood vessels, and acoustic fields affecting the threshold for blood-vessel rupture in HITU + Microbubble applications. In the experiments performed, the driving frequencies were 0.5, 1.1, 2.5, and 3.3 MHz, and the pulse repetition frequencies (PRF) were 1, 3 and 10 Hz. The duty factor was held at 0.1%. The outcomes of these *in-vivo* experiments are expected to assist in predicting the rupture probability for HITU + Microbubble procedures. They will also inform a computational model of bubble-induced vessel rupture.

**1:50**

**5pBAa3. Investigation of ultrasound exposure in the presence of microbubbles on a LumeNEXT blood-brain barrier model.** Jenna Osborn (Mech. and Aerosp. Eng., George Washington Univ., Ste. 3000, 800 22nd St. NW, Washington, DC 20052, jennakosborn@gwu.edu), Sara Bender-Bier (Food and Drug Administration, Silver Spring, MD), Asis Lopez (Food and Drug Administration, New Orleans, LA), Loren Suite, Johnny Lam, Kyung Sung, Shelby Skoog (Food and Drug Administration, Silver Spring, MD), Kausik Sarkar (Mech. and Aerosp. Eng., George Washington Univ., Washington, DC), Matthew R. Myers, and Gregory T. Clement (Food and Drug Administration, Silver Spring, MD)

The blood-brain barrier (BBB) is notoriously difficult to penetrate for the delivery of drugs and therapeutic compounds due to the unique nature of the tight junctions between endothelial cells. Ultrasound in the presence of microbubbles has been discovered to transiently open the BBB, however, little is known about the mechanism and the effects of excitation parameters at a cellular level. In this study, a static vascular lumen *in vitro* model, LumeNEXT, was utilized to investigate the excitation parameters in a controlled manner. Fibrin lumen hydrogels were lined with human cerebral microvessel endothelial and astrocyte cells. Cells were exposed to ultrasound with and without the presence of microbubbles. The devices were exposed to ultrasound parameters ranging in frequency from 0.5 to 1.55 MHz, duty factor from 0.0001–0.001 and pulse repetition frequency of 1–10 Hz. Acoustic and thermal characterization of the device was performed. Effects of microbubble size and composition on the endothelial permeability were also investigated. The transepithelial electrical resistance (TEER) and cell morphology were monitored post-ultrasound exposure to investigate endothelial permeability duration and extent. Understanding the fundamentals of ultrasound-induced permeability in a controlled manner can lead to further investigations and optimization for *in vivo* and clinical settings.

**2:10**

**5pBAa4. Ultrasound in the presence of microbubbles in a dynamic *in vitro* microfluidic blood-brain barrier model.** Jenna Osborn (Mech. and Aerosp. Eng., George Washington Univ., Ste. 3000, 800 22nd St. NW, Washington, DC 20052, jennakosborn@gwu.edu), Sara Bender-Bier, Asis Lopez (Food and Drug Administration, Silver Spring, MD), Shelby Skoog (Food and Drug Administration, Silver Spring, MD), Kausik Sarkar (Mech. and Aerosp. Eng., George Washington Univ., Washington, DC), Matthew R. Myers, and Gregory T. Clement (Food and Drug Administration, Silver Spring, MD)

Controllable, localized permeability of the blood-brain barrier (BBB) has been a goal of health professionals for enhanced delivery of drugs and therapeutic compounds for cancer and other neurological medical conditions. The transient permeability induced by transcranial ultrasound exposure in the presence of microbubbles has been shown to be a promising option, but little is known about the exact mechanism and response on a cellular level. A microfluidic device developed to recreate the *in vivo* microenvironment, including physiological fluid flow, was seeded with human cerebral microvessel endothelial cells surrounded by a matrix of astrocytes. The devices were exposed to ultrasound with and without the presence of microbubbles. The ultrasound parameters ranged in frequency from 0.5 to 1.55 MHz, duty factor from 0.0001–0.001 and pulse repetition frequency of 1 Hz–10 Hz. Microbubble size, composition, and concentration within the pumped fluid were also varied, along with varied flow rate. The transepithelial electrical resistance (TEER) and cell morphology were monitored throughout and post ultrasound exposure to understand the effects on permeability and cell viability. With this controlled variation of the ultrasound and microbubble parameters, a greater understanding of the induced permeability could lead to more success for clinical utilization.

## Session 5pBAb

## Biomedical Acoustics: General Biomedical Acoustics: Therapeutics IV

John M. Cormack, Chair
*Department of Medicine, University of Pittsburgh, Pittsburgh, PA 15261*

**Chair's Introduction—2:50**

### *Contributed Papers*

**2:55**

**5pBAb1. Effects of shell chemistry on acoustic response and dissolution behavior of lipid-coated microbubbles.** Roozbeh Hassanzadeh Azami (Mech. and Aerosp. Eng., The George Washington Univ., 800 22nd St. NW, Ste. 3000, Washington, DC 20052, roozbehazami@gwu.edu), Mitra Aliabouzar (Univ. of Michigan, Ann Arbor, MI), Jenna Osborn, and Kausik Sarkar (Mech. and Aerosp. Eng., The George Washington Univ., Washington, DC)

Microbubbles have been widely used as contrast agents for ultrasound imaging. They are also being investigated for therapeutic applications including as a vehicle for ultrasound activated targeted drug delivery. Here the effects of the shell chemistry on the acoustic response and dissolution behavior of microbubbles have been investigated. Microbubbles encapsulated by a lipid shell (DPPC and DPPE-PEG2000) were prepared with different PEG molar ratios (0% to 20%) investigating their concentration, size distribution, stability and material properties. Note that different PEG concentrations were previously shown to result in different PEG surface configuration: mushroom for lower PEG concentrations and brush for higher ones. The microbubbles in the brush regime seem to generate higher fundamental and subharmonic scattering components at 2.25 MHz excitation, while the higher harmonics seems unaffected. Microbubbles in the mushroom regime showed different dissolution behavior with longer growth period before dissolution. This is believed to be resulted from a less permeable shell in comparison to shells in brush regime. The relation between acoustic response, dissolution behavior and material properties of the shell will be discussed.

**3:15**

**5pBAb2. Acoustofluidic-mediated molecular delivery to human T cells for improved immunotherapies.** Riyakumari K. Patel (Medicine, Univ. of Louisville, 261 N Dixie Blvd, Radcliff, KY 40160, rkpate06@louisville.edu), Mariah Priddy, Emily Murphy, Bryce Stamp (Medicine, Univ. of Louisville, Louisville, KY), Connor Centner (Bioengineering, Univ. of Louisville, Louisville, KY), Paula Bates (Medicine, Univ. of Louisville, Louisville, KY), Kavitha Yaddanapudi (Surgery, Univ. of Louisville, Louisville, KY), and Jonathan A. Kopechek (Bioengineering, Univ. of Louisville, Louisville, KY)

Cell-based immunotherapies, such as CAR-T therapy, have recently demonstrated significant efficacy for treatment of blood cancers and other diseases. However, current methods of cell loading and processing for these therapies are time-consuming, expensive, and inefficient, which limits patient access and safety of this potentially life-saving therapy. To address these limitations, we are developing a novel 3-D-printed acoustofluidic system which integrates ultrasound (5 MHz) and a flow chamber to consistently induce molecular delivery to cells as they pass through the system sequentially. We conducted experimental studies to evaluate the efficiency of molecular delivery to human Jurkat T cells using the acoustofluidic system in combination with ultrasound contrast agents. Intracellular delivery of a fluorescent compound, calcein, was measured via flow cytometry. Acoustofluidic treatment significantly enhanced calcein delivery to Jurkat T cells by $51 \pm 27\%$ compared to untreated control groups without ultrasound exposure ($p = 0.005$). Cell viability, as measured with propidium iodide staining, was not compromised by acoustofluidic treatment ($93 \pm 5\%$ viability). Our experiments revealed enhanced delivery of calcein to human T cells while maintaining cell viability. Further development of this platform technology may lead to improved cell loading and processing methods for cell-based immunotherapies or other applications.

**3:35**

**5pBAb3. Histotripsy of a collagen tissue-mimicking phantom.** Jacob C. Elliott (Graduate Program in Acoust., The Penn State Univ., Res. West, State College, PA 16801, jce29@psu.edu), Molly Smallcomb, Julianna Simon (Graduate Program in Acoust., The Penn State Univ., State College, PA), Meghan E. Vidt, Sujata Khandare, and Ali A. Butt (Biomedical Eng., The Penn State Univ., State College, PA)

Collagenous tissues, such as tendon, have proven resistant to mechanical fractionation by histotripsy. Evidence on B-mode ultrasound images suggests the successful creation of boiling bubbles and/or cavitation bubble clouds in these collagenous tissues; however, the oscillation and collapse of the bubbles does not result in tissue fractionation. Here, tissue-mimicking collagen gels were placed at the focus of a 1.5-MHz HIFU transducer and insonified at various pulse lengths and repetition frequencies. Cavitation activity was monitored passively with a Philips/ATL L7-4 imaging transducer (Bothell, WA USA) and Vantage® research ultrasound system (Verasonics, Kirkland, WA, USA) and compared to high-speed photographs (Photron Nova S-9, Tokyo, Japan). Preliminary results in the tissue-mimicking collagen phantoms show violent cavitation activity and rapid gel fractionation with conventional histotripsy parameters, which is dissimilar to what is observed in highly collagenous tissues. To address this limitation, we will explore phantoms with superior fiber alignment, such as fibrin gels, which will allow for evaluation of histotripsy parameters that promote mechanical fractionation in highly collagenous tissues. [Work supported by NIH R21EB027886 and NSF GRF #DGE1255832.]

**3:55**

**5pBAb4. Assessment of clot degradation under the action of histotripsy and a thrombolytic drug.** Samuel A. Hendley (Univ. of Chicago, 5812 S Ellis Ave., IB-016, Chicago, IL 60637, hendley@uchicago.edu), Jonathan Paul, and Kenneth B. Bader (Univ. of Chicago, Chicago, IL)

Deep vein thrombosis is a major source of morbidity worldwide, affecting 5% of the population. For critical obstructions, catheter-directed thrombolytics are the frontline therapy for vessel recanalization. Adjuvant therapies to lytics are under development to improve treatment efficacy and reduce procedure-related complications. One such adjuvant is histotripsy, a focused ultrasound therapy that relies on the nucleation of bubble cloud to modulate tissue. This combination approach has been successful *in vitro*, and is hypothesized to promote clot dissolution via two mechanisms: mechanical hemolysis and enhanced fibrinolysis. In this study, the contributions of hemolysis and fibrinolysis to clot degradation under histotripsy and

a thrombolytic were quantified with measurements of hemoglobin and D-dimer, respectively. Linear regression analysis was used to determine the relationship between hemoglobin, D-dimer, and the overall treatment efficacy (clot mass loss). A similar analysis was conducted to gauge the contribution of bubble activity assessed with passive cavitation imaging on hemolysis and fibrinolysis. Tabulation of these data demonstrated hemolysis and fibrinolysis contributed equally to clot mass loss. Furthermore, bubble cloud activity promoted the generation of hemoglobin and D-dimer in equal proportion. These studies indicate a multifactorial process for clot degradation under the action of histotripsy and a lytic therapy.

FRIDAY AFTERNOON, 11 DECEMBER 2020 1:05 P.M. TO 2:35 P.M.

## Session 5pNSa

## Noise and Psychological and Physiological Acoustics: Advances in Hearing Protection Devices III

Cameron J. Fackler, Cochair
*3M Personal Safety Division, 7911 Zionsville Rd., Indianapolis, IN 46268*

William J. Murphy, Cochair
*Division of Field Studies and Engineering, Noise and Bioacoustics Team, National Institute for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998*

Elliott H. Berger, Cochair
*Berger Acoustical Consulting, 221 Olde Mill Cove, Indianapolis, IN 46260*

**Chair's Introduction—1:05**

### *Invited Papers*

**1:10**

**5pNSa1. Comparison of hearing protector ratings for statistically significant differences.** William J. Murphy (Div. of Field Studies and Eng., Noise and Bioacoustics Team, National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov)

During the development of the American National Standards Institute S12.68 standard for estimating effective A-weighted sound pressure levels when hearing protector device (HPD) are worn, the need to provide a mechanism to compare HPD ratings was identified. The current Noise Reduction Rating (NRR) is a single number that describes the potential of a product to reduce noise at the user's ears. The Noise Reduction Statistic for A-weighting ($NRS_A$) describes a range of protection performance and estimates the uncertainty associated with the rating. The uncertainty for $NRS_A$ is estimated using a bootstrap procedure. This method can be applied to current ratings and comparisons between rating evaluations from different laboratories. This paper will discuss NRR and $NRS_A$ error bars ratings and present a modification of the bootstrap method to estimate directly the statistical significance of differences. Results from two different studies of hearing protector attenuation measurements and training protocols will be examined. The studies' results will also be compared against the manufacturers' published attenuation values.

**1:30**

**5pNSa2. Susceptibility to impulsive noise: A review of early studies.** Gregory Flamme (SASRAC, 2264 Heather Way, Forest Grove, OR 97116, gflamme@sasrac.com), Kristy K. Deiters, and Stephen M. Tasko (SASRAC, Forest Grove, OR)

The maximum tolerable exposures returned by damage-risk criteria for impulsive noise are often designed to protect the population to the 5th percentile most susceptible exposed person. To protect human research volunteers, estimates of this location on the distribution must be estimated rather than measured directly. In this presentation, we present threshold shift distributions from human studies of threshold shifts prior to about 1980. Sample sizes ranged in these studies ranged between 5 and 66 ears. The threshold shift distributions implied by these data are compared against assumptions made in more recent damage-risk criteria for impulsive noise. Implications for the development and/or improvement of damage-risk criteria will be discussed.

**1:50**

**5pNSa3. Significance of middle ear model for prediction of metabolic exhaustion of the auditory system.** Brissi Zagadou (L3Harris Technologies, Inc., 10180 Barnes Canyon Rd., San Diego, CA 92121, Brissi.Zagadou@L3Harris.com)

Modern day military noise is complex, involving irregular impulses riding on top of moderate continuous background noise. An accurate prediction of injury from complex noise exposure requires both a cochlear metabolic exhaustion (ME) model and an accurate middle ear model (MEM), accounting for acoustic reflex (AR) and annular ligament (AL) nonlinearities. An end-to-end biomechanical model including a ME model was developed that established a good correlation between the outer hair cell energy deficit (OHC-D) outcomes and the temporary threshold shift (TTS) data from chinchilla complex noise exposure, explained the data collected, and provided insights on the effect of complex noise exposure on outcomes, including the effect of inter-pulse Interval (IPI), shot sequence, and background noise. Literature data were used to derive the open ear and middle ear transfer functions (TF). In this paper, the model results are discussed with an emphasis on the MEM component. The differences in the end-to-end model results are compared between the new MEM and a MEM adopted from the improved AHAAH-ICE model that includes the AR and AL nonlinearities, with model parameters adjusted for chinchilla. The results show that the TFs from the two MEMs cannot be matched using mere parametric optimization, due mainly to the fact that the chinchilla middle ear is rather tuned to the higher frequencies compared to that of human. There is a need for an improved MEM for the chinchilla middle ear TF including AR effects.

**2:10**

**5pNSa4. In-ear noise dosimetry in an industrial environment.** Kevin Michael (Michael & Assoc., Inc., 2766 W. College Ave., Ste. 1, State College, PA 16801, kevin@michaelassociates.com)

In-ear noise dosimetry is a new, promising method of preventing noise-induced hearing loss (NIHL). Specifically, when in-ear noise dosimetry is used on a daily basis, a complete noise exposure history is established for an employee. With this in hand, the hearing conservationist can monitor exposures and intervene when necessary, taking whatever steps are required to prevent future overexposures. These steps can include changing protective devices, increasing wearing duration and improving fitting techniques. Since NIHL generally occurs after months and years of overexposures, the immediate intervention will prevent the hearing loss from progressing. Because the damage-risk criteria were developed using a sound level meter in open space, accurate in-ear measurements must compensate for the transfer function of the open ear (TFOE). This is challenging due to the variability of TFOE across the population and due to various microphone locations within the ear canal. A ‘generic’ TFOE can be used as an estimate for all users, or the TFOE can be measured on each employee by comparing an in-ear microphone measurement and a sound field microphone measurement of the same stimulus. These approaches will be discussed as fundamental elements of an industrial in-ear noise dosimetry program.

## Session 5pNSb

# Noise and Psychological and Physiological Acoustics: Advances in Hearing Protection Devices IV

Cameron J. Fackler, Cochair
*3M Personal Safety Division, 7911 Zionsville Rd., Indianapolis, IN 46268*

William J. Murphy, Cochair
*Division of Field Studies and Engineering, Noise and Bioacoustics Team, National Institute for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998*

Elliott H. Berger, Cochair
*Berger Acoustical Consulting, 221 Olde Mill Cove, Indianapolis, IN 46260*

**Chair's Introduction—2:50**

## *Invited Papers*

**2:55**

**5pNSb1. Testing the fit of a new standard for hearing protection fit testing.** Cameron J. Fackler (3M Personal Safety Div., 7911 Zionsville Rd., Indianapolis, IN 46268, cameron.fackler@mmm.com) and Lauraine L. Wells (3M Co., St. Paul, MN)

Systems that test the fit of hearing protection devices (HPDs) on individual users are an increasingly popular tool in hearing conservation. ANSI/ASA S12.71-2018, *American National Standard Performance Criteria for Systems that Estimate the Attenuation of Passive Hearing Protectors for Individual Users*, is the first standard worldwide for HPD fit-test systems, referred to as field attenuation estimation systems (FAESs). The S12.71 standard specifies a variety of system performance requirements that may be voluntarily adopted by the manufacturer of a FAES. It also specifies experimental procedures to evaluate FAES performance and uncertainty. This presentation gives an overview of the requirements outlined in S12.71-2018 and shares our experiences in evaluating a commercially available FAES for compliance with this standard.

**3:15**

**5pNSb2. Image-based estimation of hearing protection attenuation and fit.** Christopher J. Smalt (Human Health & Performance Systems, MIT Lincoln Lab., 244 Wood St., Lexington, MA 02420, Christopher.Smalt@ll.mit.edu), Gregory Ciccarelli, Aaron Rodriguez (Human Health & Performance Systems, MIT Lincoln Lab., Lexington, MA), and William J. Murphy (Div. of Field Studies and Eng., Noise and Bioacoustics Team, National Inst. for Occupational Safety and Health, Cincinnati, OH)

Dangerous acoustic noise levels are encountered occupationally by 22 million workers annually (Tak *et al*., 2009). Occupational and recreational noise is known to cause permanent hearing damage and reduced quality of life, which suggests the importance of noise controls including hearing protection devices (HPDs). While HPDs can provide adequate protection for many noise exposures, it is often a challenge to properly train and maintain compliance. Fit-testing systems are commercially available to ensure proper attenuation is achieved, but they either require specific facilities (e.g., ANSI S12.6) or special equipment and hearing protection devices designed for microphone-in-the-ear (MIRE) testing. In this study, we estimated the adequacy of fit for a foam hearing protector using only a photograph of the inserted foam plug. Our initial dataset was photographs of a single type of foam hearing protector, broken up into a 200-image training set and 40 image test set, where individual subjects were not mixed across the two datasets. We achieved 73% classification accuracy if the fit was greater or less than the median measured attenuation (35 dB averaged across frequency). Ultimately, this algorithm could be used as part of a smartphone app for training as well as for automated compliance monitoring in noisy environments for preventing hearing loss.

**3:35**

**5pNSb3. Validating a hearing protector with built-in sound level measurement.** Jackie Di Francesco (Honeywell, 7828 Waterville Rd., San Diego, CA 92154, jacqueline.difrancesco@honeywell.com)

Personal noise measurement, or dosimetry, is a useful tool for estimating a worker's noise exposure over time. The data collected can be used to mitigate risk of hearing loss and other negative health effects of hazardous noise. It can also be used to ensure compliance with hearing conservation regulations and policies. Conventional noise dosimetry typically measures environmental noise levels from the location of the wearer's shoulder. Measuring at the shoulder, however, does not take into account the attenuation of any hearing protection that is used. A hearing protector with a built-in sound level measurement system can measure the actual noise levels beneath the hearing protector, as well as the surrounding noise. Due to COVID-19 restrictions, a field study to validate hearing protector measurements in real-world environments could not be completed. This study uses alternative methods to validate the accuracy of the hearing protector's measurements in different noise environments.

## *Contributed Paper*

3:55

**5pNSb4. Development of a tablet-based fit test system for military and austere environments.** Devon Kulinski (Audiol. and Speech Pathol., Walter Reed NMMC, 4954 North Palmer Rd., BLDG 19-5500, Bethesda, MD 20889, devon.m.kulinski.ctr@mail.mil), Matthew J. Makashay (Army Hearing Program, U.S. Army Public Health Ctr., Aberdeen Proving Ground, MD), Coral Dirks (Audiol. and Speech Pathol., Walter Reed NMMC, Bethesda, MD), Benjamin Sheffield (Army Hearing Program, U.S. Army Public Health Ctr., Bethesda, MD), and Douglas S. Brungart (Audiol. and Speech Pathol., Walter Reed NMMC, Bethesda, MD)

Hearing protection device (HPD) fit testing is considered “best practice” for hearing loss prevention programs. While DoD regulations recommend fit testing for Service Members (SMs) who experience a threshold shift, current field attenuation estimation systems (FAES) are not readily accessible to many hearing conservation personnel, in part due to issues with portability, cost and scalability. In this study, we compare the efficacy of several novel fit-testing paradigms against an objective field-microphone in real ear (F-MIRE) system. The paradigms include a real-ear attenuation at threshold (REAT) protocol using a compact headphone-based boothless audiometer, a broadband “HPD Check” screener using a headphone-based boothless audiometer, and a loudness balancing (LB) protocol using low-cost headphones. All three paradigms operate via a commercial android tablet. The low-cost tablet-based protocol in particular might make it feasible to conduct fit testing on all SMs, and not just those who experience significant hearing changes or who are being fitted with hearing protection for the first time. The enhanced portability of the FAES described here would also offer advantages outside of the clinic to ensure proper HPD fit prior to noise-hazardous weapons system exposures in the military training environment. [The views expressed here are those of the author and do not reflect the official policy of the Department of Army, Department of Defense, or U.S. Government.]

FRIDAY AFTERNOON, 11 DECEMBER 2020 — 1:05 P.M. TO 2:35 P.M.

# Session 5pPAa

## Physical Acoustics: General Topics: Potpourri III

Curtis Rasmussen, Cochair
*University of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78712*

Matthew S. Byrne, Cochair
*Electrical and Computer Engineering, University of Texas at Austin, 2501 Speedway, Room 5.838-A (Seat 5), Austin, TX 78751*

**Chair’s Introduction—1:05**

## *Contributed Papers*

1:10

**5pPAa1. Chu limit and non-Foster radiation for sound.** Curtis Rasmussen (Univ. of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78712, curtis.wiederhold@gmail.com) and Andrea Alù (Photonics Initiative, CUNY Adv. Sci. Res. Ctr., New York, NY)

We show that the quality factor of a small passive acoustic radiator is fundamentally limited by its volume normalized to the emitted wavelength, imposing important constraints on the bandwidth and efficiency of compact acoustic sources. We overcome this bound, known in electromagnetics as Chu’s limit, by realizing a piezoelectric transducer loaded with a non-Foster circuit, showing that its radiation bandwidth is fundamentally limited only by stability considerations. Based on these principles, we experimentally observe over a threefold bandwidth enhancement compared to its passive counterpart, paving the way towards non-Foster acoustic elements and metamaterials that overcome the bandwidth constraints of passive systems.

1:30

**5pPAa2. Broadband phononic frequency comb generation in a parametrically excited capacitive micromachined ultrasonic transducer.** Sushruta Surappa (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, Atlanta, GA 30313, sushsurappa@gatech.edu) and F. Levent Degertekin (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Optical frequency combs consisting of evenly spaced discrete frequencies have been used for a number of applications such as frequency metrology, range finding and molecular fingerprinting. Recently, the acoustic equivalent of these frequency combs, known as phononic frequency combs (PFCs), have been demonstrated in micro and nanoscale resonators. However, such demonstrations are restricted to PFC generation in high-Q mechanical resonators under carefully controlled operating conditions with limited practical applications. In this work, we demonstrate a novel method of generating PFCs with a parametrically driven capacitive micromachined ultrasonic transducer (CMUT) operating in a fluid medium. The proposed system consists of an electrically driven CMUT array that forms part of a

resonant RLC circuit of frequency $w_0$. By applying two drive tones of frequency $w_0$ and $w_0+\Delta$, we are able to generate broadband acoustic frequency combs via a parametric four-wave mixing process. The intensity and number of combs generated is strongly dependent on the relative strength of the two driving tones whereas the comb spacing is determined by the frequency difference $\Delta$ between the drive signals. We also briefly discuss a potential application of PFCs in extended non-ambiguous range (NAR) distance metrology with interferometric resolution.

**1:50**

**5pPAa3. Far-field acoustic subwavelength imaging with structured illumination.** Jinuan Lin (Elec. and Comput. Eng., Univ. of Wisconsin-Madison, 1415 Eng. Dr., Rm. 3533, Madison, WI 53706, jlin328@wisc.edu) and Chu Ma (Elec. and Comput. Eng., Univ. of Wisconsin-Madison, Madison, WI)

In the past decade, subwavelength imaging technologies have attracted a lot of interest for breaking the diffraction limit. While localization-based methods are able to image isolated subwavelength objects, differentiating objects with subwavelength distances in the far field is still challenging. To solve this problem, it has been proposed in optical microscopy to exploit structured or blind structured illumination, where the spatial frequency mixing between the objects and the illumination converts evanescent waves to propagating waves that can reach far field. Here, we propose an acoustic analogy and demonstrate a framework, both theoretically and experimentally, for performing far-field acoustic subwavelength imaging using structured illumination generated by scattering media with subwavelength features. A scattering medium is placed behind subwavelength objects. The excitation wave is first diffracted by the scattering medium before it further passes through the objects and reaches the receiver in the far field. By utilizing a compressive sensing reconstruction algorithm, the image of the objects can be reconstructed with multiple measurements that are obtained by shifting or rotating the scattering medium. The proposed framework has great potential in medical imaging, non-destructive testing, and underwater acoustic imaging for improving imaging resolution of objects far away from the sensors.

**2:10**

**5pPAa4. Acoustic wave propagation and reciprocity in a toroidal waveguide carrying a mean flow.** Charles Thompson (Elec. and Comput. Eng, UMASS Lowell, 1 Univ. Ave., Lowell, MA 01854, charles_thompson@uml.edu), Sarah Kamal, Lejun Hu, and Kavitha Chandra (Elec. and Comput. Eng, UMASS Lowell, Lowell, MA)

In this paper acoustic wave propagation in a toroidal waveguide of rectangular cross-section carrying a circumferentially directed mean flow is analyzed. The typical length scale of variation of the pressure is given by the circumference $L_0$ , the height of the waveguide $H_0$ and the gap $\Delta$R$. Of particular interest are cases where $\varepsilon = H_0/L_0$ is much less than one. It is shown that flow reverse symmetry of the acoustic pressure gives rise to nonreciprocal behavior between select points along the toroid's circumference.

FRIDAY AFTERNOON, 11 DECEMBER 2020 — 2:50 P.M. TO 4:00 P.M.

## Session 5pPAb

## Physical Acoustics: General Topics: Potpourri IV

Matthew S. Byrne, Cochair
*Electrical and Computer Engineering, University of Texas at Austin, 2501 Speedway, Room 5.838-A (Seat 5), Austin, TX 78751*

Curtis Rasmussen, Cochair
*University of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78712*

**Chair's Introduction—2:50**

### *Contributed Papers*

**2:55**

**5pPAb1. Propagation of vortex beams in a stratified medium.** Xudong FAN (Univ. of MS, 145 Hill Dr., University, MS 38677, xfan1@go.olemiss.edu), Zheguang Zou (Dept. of Phys. and Astronomy, Univ. of MS, University, MS), and Likun Zhang (Univ. of MS, Oxford, MS)

Wave vortices, with a spiral phase proportional to azimuth angle, can be used for particle manipulations and underwater navigation. Prior studies focused on acoustic vortices in homogeneous media, however, media like tissues or ocean environments can be inhomogeneous. Here, the propagation of vortex beams in a linearly stratified medium, where the sound speed is a function of depth, is simulated. The evolution of the vortex fields and linear/angular momentum are examined to reveal the interaction of the stratification and the vortex beams. A series of dynamic behaviors have been observed including distorting of the fields and reversal of angular momentum [Fan *et al.*, *Phys. Rev. Res.* **1**(3), 032014 (2019)]. The complexity of vortex beam propagation in inhomogeneous media and inspired applications are suggested.

**3:15**

**5pPAb2. Point to point propagation over a phase gradient grooved surface.** Steve Mellish (Eng. and Innovation, STEM, The Open Univ., Milton Keynes MK7 6AA, United Kingdom, steve@melectronics.co.uk), Shahram Taherzadeh, and Keith Attenborough (Eng. and Innovation, STEM, The Open Univ., Milton Keynes, United Kingdom)

Previous work on deriving a modal theory for predicting point-to-point propagation over a uniformly grooved surface is extended to predict propagation over a periodic variable depth grooved surface with a low flow resistivity porous material of different thickness in each groove. The form of the surface is intended to create an approximately linear phase gradient for plane waves reflected from the surface. The resulting attenuation due to

ground effect is found to be larger and more broadband than predicted over either an acoustically hard grooved surface or an acoustically soft layer with thickness equal to the largest groove depth for the same source-receiver geometry. Various contributions to the excess attenuation spectrum from surface waves associated with the different groove depths, quarter wavelength resonances in the grooves, the phase gradient for plane wave reflections and diffracted modes are investigated. As result the broadband nature of the excess attenuation is explained and important parameters in designing such surfaces are identified.

**3:35**

**5pPAb3. Animating sound using neurally multiplexed holograms.** Athanasios G. Athanassiadis (Max Planck Inst. for Intelligent Systems, Heisenbergstr. 3, Stuttgart 70569, Germany, thanasi@is.mpg.de), Lennart Schlieder (Max Planck Inst. for Intelligent Systems, Tübingen, Germany), Kai Melde (Max Planck Inst. for Intelligent Systems, Stuttgart, Germany), Valentin Volchkov (Max Planck Inst. for Intelligent Systems, Tübingen, Germany), and Peer Fischer (Max Planck Inst. for Intelligent Systems, Stuttgart, Germany)

Acoustic holograms are a simple yet powerful tool to project structured pressure fields for applications including acoustic manipulation, ultrasonic therapy, and compressed sensing. However current holographic projectors lack the ability to dynamically change the pressure field, which is necessary for real-time manipulation and control of acoustically excited systems. A promising solution is to use hybrid projectors: multi-element phased arrays combined with high-resolution static holograms. Such a system was recently built to dynamically translate high-resolution pressure fields for particle manipulation in water. Here, we extend this paradigm and introduce a more general hybrid acoustic projector capable of projecting distinct high-resolution pressure fields using arrays with only a few elements. Each input element excites a distinct output field through a multi-plane hologram, whose phase delays are calculated by representing pixels in each phase plate as neurons in a diffractive neural network. This representation allows us to efficiently multiplex the holograms so that all of the input-output transformations are performed by a single set of static phase plates. We experimentally characterize the output of our device and show that by cycling through the different inputs, the projected acoustic field can be changed in real time, creating an "acoustic animation."

FRIDAY AFTERNOON, 11 DECEMBER 2020 — 1:05 P.M. TO 1:50 P.M.

## Session 5pSCa

## Speech Communication: Speech Corpora and Modeling (Poster Session)

Authors will be at their posters from 1:05 p.m. to 1:50 p.m.

### *Contributed Papers*

**5pSCa1. How do words compete? Quantifying lexical competition with acoustic distance.** Matthew C. Kelley (Linguist, Univ. of AB, University of AB, 3-24 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, mckelley@ualberta.ca) and Benjamin V. Tucker (Linguist, Univ. of AB, Edmonton, AB, Canada)

Research on speech perception and lexical access often uses the activation and competition metaphor to describe the process of spoken word recognition. One way of expressing competition associated with a given word is its phonological neighborhood density, which is a calculation of similarity. The present study uses acoustic distance as an alternative to phonological neighborhood density to measure lexical competition during speech perception. The quantification of a word's lexical competition is given by what is termed its acoustic distinctiveness, which is taken as its average acoustic distance to all other words in the lexicon. A variety of possible abstract acoustic representations for items in the lexicon are analyzed. Statistical modeling shows that acoustic distinctiveness has a similar effect as phonological neighborhood density. Additionally, acoustic distinctiveness consistently increases model fitness more than phonological neighborhood density, regardless of the abstract representation used. Acoustic distinctiveness, however, does not explain all the same things as phonological neighborhood density. Potential theoretical implications of acoustic distinctiveness's usefulness in statistical and psycholinguistic models are discussed.

**5pSCa2. The massive auditory lexical decision database: Acoustic analyses of a large-scale, single speaker corpus.** Ryan G. Podlubny (Linguist, Univ. of AB, 4/162 Chester St. East, Christchurch, Canterbury 8011, New Zealand, ryan.podlubny@pg.canterbury.ac.nz) and Benjamin V. Tucker (Linguist, Univ. of AB, Edmonton, AB, Canada)

Studying speech production has allowed the discovery of numerous patterns that hold within—and, at times, differ across—speaker groups (e.g., voice-onset-time as a cue for consonant voicing). However, such work rarely explores the variation and consistencies that might be observed within comparable numbers of productions from a single speaker. In fact, acoustic analyses of speech production most often survey relatively few observations from each speaker, where high participant numbers are required to offset limited by-speaker contributions. Therefore, previous research often addresses inter-speaker variation, but by nature disallows any meaningful exploration of variation/consistency across productions from a single speaker. In tandem with the Massive Auditory Lexical Decision (MALD) study [Tucker *et al.*, *Behav. Res.* **51**(3), 1187–1204 (2019)], the present work serves two purposes: (1) to analyze and describe speech stimuli comprising the MALD single-speaker corpus (26 793 English words and 9592 English pseudowords, roughly 6 h of recorded speech in total), and (2) to suggest the breadth of work possible using such a corpus. As examples of the latter, analyses are provided to compare and contrast acoustic characteristics associated with varying degrees of lexical stress, and to explore the vowel space as realized in words versus pseudowords.

**5pSCa3. Re-testing theories of vowel inherent spectral change.** Jonathan Jibson (Univ. of Wisconsin–Madison, 7134 Helen C White Hall, 600 N Park St., Madison, WI 53711, jibson@wisc.edu)

Three basic models of vowel inherent spectral change have been proposed: onset + offset, onset + direction, and onset + slope (or spectral rate of change). Morrison (2013) presents three theoretical stimuli whose identification rates relative to each other should lend weight to one of these hypotheses: (i) canonical duration and canonical onset-to-offset trajectory, (ii) canonical duration with double trajectory length, and (iii) canonical trajectory with half duration. The bulk of previous work supports onset + offset. In the present study, eight English monophthongs were synthesized with the Klatt synthesizer of the Berkeley Phonetics Machine. Three versions of each vowel were synthesized, corresponding to the theoretical stimuli of Morrison (2013). These 24 stimuli were used in two studies ($n = 18$ for each). First was a vowel identification task, where the choices were [hVd] words with the chosen vowels (*heed*, *hid*, etc.). Second was a goodness rating task with a 7-point Likert scale, where participants were played a vowel stimulus while being shown the orthographic [hVd] word containing that vowel. The results align most closely with the onset + slope hypothesis, though not as neatly as predicted.

**5pSCa4. Cue-dependent processing in an automatic acoustic cue analysis system for detecting atypical speech.** Fjona Parllaku (Speech Commun. Group, Res. Lab. of Electronics, Massachusetts Inst. of Technol., 362 Memorial, Cambridge, MA 02139, fjonap12@mit.edu), Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (Speech Commun. Group, Res. Lab. of Electronics, Massachusetts Inst. of Technol., Cambridge, MA)

Current automatic speech recognition systems have been shown to be quite accurate in handling the way machines respond to human language spoken by typical healthy speakers. However, there are still many challenges to overcome in applying these systems to the task of detecting atypically produced speech. Analysis of acoustic cue production patterns by speakers can provide more detailed information than analysis of word, phoneme or phone error rates. A system for automatic detection of acoustic cues is being developed that extracts the individual cues to distinctive features. These acoustic cues include landmark acoustic cues (vowel, glide, consonant closures and releases) as well as vowel place, consonant place, nasalization and glottal cues. We describe a cue-dependent processing framework for detecting a subset of these cues, involving smoothing, peak detection, and segmentation.

**5pSCa5. Labeling databases with individual acoustic cues to distinctive features.** Jeung-Yoon Choi (Speech Commun. Group, Res. Lab. of Electronics, MIT, 50 Vassar St., Rm 36-761, Cambridge, MA 02139, jyechoi@mit.edu) and Stefanie Shattuck-Hufnagel (Speech Commun. Group, Res. Lab. of Electronics, MIT, Cambridge, MA)

Several speech databases have been manually annotated for individual acoustic cues to distinctive features. The acoustic cue labels include 8 landmark types (Stevens 2002) related to the manner features, and 32 other types related to place and voicing features. The labeled data include isolated words and syllables, read speech and task-driven conversational speech. The isolated words and syllables are drawn from a VCV database, which consists of about 400 vowel-consonant-vowel utterances (3 speakers), and from the UConn Isolated Words dataset, which comprises around 1200 monosyllabic words each spoken by 4 speakers. The continuous speech samples include a subset (40 speakers) of the TIMIT read sentences database, and a Map Task spontaneous speech database (8 speakers) consisting of 16 conversations. These feature-cue-labeled databases can serve as a training set for cue-recognition algorithms, and can provide material for analysis of systematic context-governed cue modification patterns, such as the loss of closure and release landmarks for coda nasals with preservation of nasality in the preceding vowel, and the loss of coda stop landmarks with preservation of duration cues to voicing in the preceding vowel. In addition, an online tutorial that outlines how to label individual acoustic cues to distinctive features is under development.

**5pSCa6. Speech recognition of spoken Italian based on detection of landmarks and other acoustic cues to distinctive features.** Maria-Gabriella Di Benedetto (Radcliffe Inst. for Adv. Study at Harvard Univ., Cambridge, MA;, and Sapienza Univ. of Rome, Rome, Italy., Via Eudossiana 18, Rome 00184, Italy, mariagabriella.dibenedetto@uniroma1.it), Jeung-Yoon Choi, Stefanie Shattuck-Hufnagel (MIT, Cambridge, MA), Luca De Nardis, Sara Budoni, Jacopo Vivaldi (Sapienza Univ. of Rome, Rome, Italy), Javier Arango, Alec DeCaprio, and Stephanie Yao (Radcliffe Inst. for Adv. Study at Harvard Univ., Cambridge, MA)

Modeling the process that a listener actuates in deriving words intended by a speaker, requires setting a hypothesis on how lexical items are stored in memory. Stevens' model (2002) postulates that lexical items are stored in memory according to distinctive features, and that these features are hierarchically organized. The model highlights the importance of abrupt acoustic events, named landmarks, in the perception process. In this model, the detection of landmarks is primary in human perception, corresponding to the first phase of recognition. The temporal area around the landmark is then further processed by the listener. Based on the above model, the Speech Communication Group of the Massachusetts Institute of Technology (MIT) developed a speech recognition system—for spoken English—over a span of more than 20 years. In the current work (LaMIT project, Lexical access Model for Italian) the above model is applied to Italian. Exploring a new language will provide insight into how Stevens' approach has universal application across languages, with relevant implications for understanding how the human brain recognizes speech. K. N. Stevens "Toward a model for lexical access based on acoustic landmarks and distinctive features," *J. Acoust. Soc. Am.*, **111**(4), 1872–1891 (2002).

**5pSCa7. Phonological processes of consonants from orthographic to pronounced words in the Buckeye Corpus.** Byunggon Yang (English Education Dept., Pusan National Univ., Pusantehakro63-2, Keumjunggu, Pusan 46241, South Korea, byunggonyang@gmail.com)

The phonological processes of consonants in pronounced words in the Buckeye Corpus are investigated along with the frequency distribution of these processes to provide a clearer understanding of conversational English for linguists and teachers. Both orthographic and pronounced words were extracted from the transcribed label scripts of the Buckeye Corpus. Next, the phonological processes of consonants in the orthographic and pronounced labels were tabulated separately by onsets and codas, and a frequency distribution by consonant process types was examined. The results showed that the majority of the onset clusters were pronounced as the same sounds in the Buckeye Corpus. The participants in the corpus were presumed to speak semiformally. In addition, the onsets have fewer deletions than the codas, which might be related to the information weight of the syllable components. Moreover, there is a significant association and strong positive correlation between the phonological processes of the onsets and codas in men and women. We conclude that an analysis of phonological processes in spontaneous speech corpora can contribute to a practical understanding of spoken English. Further studies comparing the current phonological process data with those of other languages would be desirable to establish universal patterns in phonological processes.

**5pSCa8. Automatic detection of t/d deletion using forced alignment.** Lisa Lipani (Linguist Dept., Univ. of Georgia, 142 Gilbert Hall, Athens, GA 30602, llipani@uga.edu)

In phonetic research, the time needed to manually annotate large-scale data is often prohibitive, and computer-assisted alignment is needed. Forced alignment, an offshoot of automatic speech recognition, is a technique that provides word and phone boundaries of use to linguists. However, this forced alignment systems rely on a dictionary that typically gives canonical pronunciations of words. This study investigates the efficacy of modifying the dictionary of the Montreal Forced Aligner (McAuliffe *et al.*, 2017) to account for a well-attested sociophonetic variation phenomenon, t/d deletion. This is done by aligning the Buckeye Corpus (Kiesling *et al.*, 2006), chosen as it allows comparison of forced alignment results to human

transcriber results. Overall, 23 522 words that canonically feature word-final t/d in a consonant cluster were examined. Forced alignment results from this modification were in agreement with human transcribers approximately 71% of the time, close to the 76% agreement of human transcribers (Raymond *et al.*, 2002). These results are promising for future large-scale sociophonetic research in which dictionary modifications can be made to better force align data containing sociophonetic variation.

**5pSCa9. Estimation of the frequency of occurence of italian phonemes in text.** Javier Arango (Radcliffe Inst. for Adv. Study at Harvard Univ., 10 Garden St., Cambridge, MA 02138, jarango@college.harvard.edu), Alec DeCaprio, Stephanie Yao (Radcliffe Inst. for Adv. Study at Harvard Univ., Cambridge, MA), Sunwoo Baik, Stefanie Shattuck-Hufnagel (Massachusetts Inst. of Technol. (MIT), Cambridge, MA), and Maria-Gabriella Di Benedetto (DIET, Sapienza Univ. of Rome, Rome, Italy)

The purpose of this project was to derive a reliable estimate of the frequency of occurrence of the 30 phonemes – plus consonant geminated counterparts- of the Italian language, based on four selected written texts. Since no comparable dataset was found in previous literature, the present analysis may serve as a reference in future studies. Four textual sources were considered: Come si fa una tesi di laurea: le materie umanistiche by Umberto Eco, I promessi sposi by Alessandro Manzoni, a recent article in Corriere della Sera (a popular daily Italian newspaper), and In altre parole by Jhumpa Lahiri. The sources were chosen to represent varied genres, subject matter, time periods, and writing styles. Results of the analysis, which also included an analysis of variance, showed that, for all four sources, the frequencies of occurrence reached relatively stable values after about 6000 phonemes (approximately 1250 words), varying by $<0.025\%$. Estimated frequencies are provided for each single source and as an average across sources.

FRIDAY AFTERNOON, 11 DECEMBER 2020 — 1:50 P.M. TO 2:35 P.M.

## Session 5pSCb

## Speech Communication: Speech Production and Perception (Poster Session)

Authors will be at their posters from 1:50 p.m. to 2:35 p.m.

### *Contributed Papers*

**5pSCb1. The production and perception of prevelar/æ/-raising by Canadian and American english speakers.** Lisa Sullivan (Dept. of Linguist, Univ. of Toronto, 4th Fl. 100 St. George St., Toronto, ON M5S 3G3, Canada, lisa.sullivan@mail.utoronto.ca)

Pre-velar /æ/-raising occurs when /æ/ is raised before /g/ relative to other contexts. This study examines the extent of phonological conditioning of /æ/-raising between /g/ and /k/ in production and perception. First, I tested the extent to which 18 Canadian and American English speakers raise /æ/ before /g/ (vs. /k d t/) using a wordlist reading task with words containing /æ/ and /ɛ/ in the relevant contexts. Consistent with previous studies (Stanley, 2018; 2019), Canadians, but not Americans, tended to raise /æ/ before /g/. Second, I tested whether the same effect exists in perception using a forced-choice nonce-word identification task with 9-step continua from /æ/ to /ɛ/ before /g/ and /k/. Are speakers who /æ/-raise in production more likely to identify the manipulated vowel as /æ/ before /g/ than /k/, especially when it is ambiguous? Perception has not been tested in previous work, though anecdotal evidence suggests differences in perceptual saliency – Americans, but not Canadians, are aware of /æ/-raising, suggesting that a relationship may exist. Contrary to the anecdotal evidence, there was no correlation between production and perception at the group (nationality) or individual level. However, /æ/ was perceived more before /g/ than /k/ overall, suggesting that /æ/-raising influences perception.

**5pSCb2. Production and perception of a merger: The case of [l] and [n] in Eastern Min Chinese.** Ruoqian Cheng (Dept. of Linguist, Univ. of Kansas, 1541 Lilac Ln., Lawrence, KS 66045, rqcheng@ku.edu) and Allard Jongman (Dept. of Linguist, Univ. of Kansas, Lawrence, KS)

We identified six acoustic correlates to the phonemic contrast between word-initial [l] and [n] in both English and Mandarin (Cheng and Jongman, 2019, *JASA*). All six cues suggested the phonemic contrast between prescriptive L and N (phonemes in the unmerged system) disappeared in Eastern Min (EM) in production. The current study investigated the phonological and social factors that modulate the merger-in-progress. In production, a classifier was trained with acoustic parameters from Mandarin (F3, F1 bandwidth, A1-P0) to categorize EM tokens. Linear discriminant results illustrated acoustically distinct [l] and [n] tokens as the phonetic realizations of the merger. Both prescriptive L and N were variably realized as acoustic [l] or [n] in production. Low vowels correlated with acoustically more nasal-like onsets. Speakers of all ages and genders leaned toward [l]. Perceptually, AX discrimination results (ISI = 750 ms) suggested EM speakers did not distinguish prescriptive L from N (mean $A' = 0.46$), providing converging evidence of the merger. EM listeners did not perceive the difference between acoustic [l] and [n] either (mean $A' = 0.47$). However, lower vowels with nasal codas illustrated a higher degree of sensitivity than other phonological conditions, which was an effect of the larger acoustic distance between stimuli.

**5pSCb3. Word-final incomplete neutralization in Afrikaans: Differences in perception and production.** Alexandra M. Pfiffner (Linguist, Georgetown Univ., 1421 37th St. N.W., Poulton Hall, Washington, DC 20007, amp343@georgetown.edu)

This study examines incomplete neutralization in word-final plosive voicing in Afrikaans. Perception and production data were collected from 34 native speakers, divided into two gender-balanced age groups (20–24, 60–83). Perception task stimuli were artificially manipulated continua (/stat/-/stad/ "township," "city" and /raat/-/raad/ "folk medicine," "advice") varying in preceding vowel length, closure duration, burst duration, or a combination of cues. Participants underwent a two-alternative forced-choice task, where they heard a stimulus word and identified if they heard *stat* or *stad*, or *raat* or *raad*. Results show that only preceding vowel length was a significant factor across all participant groups: longer vowel lengths were associated with word-final underlying voicing, and shorter vowel lengths with underlying voicelessness. In the production task, participants read a randomized word list. Results show that all age and gender groups have systematic differences in preceding vowel length and burst duration. A distinction in closure duration in the predicted direction was maintained only by younger participants. The comparison of perception and production results have implications for the relative weighting of cues, and further suggest that word-final incomplete neutralization may not be a stable phenomenon in Afrikaans.

**5pSCb4. Individual differences in the production and perception of prosodic boundaries in American English.** Jiseung Kim (Linguist, Univ. of Michigan, 416 S 5th Ave., Apt 4, Ann Arbor, MI 48104, jiseungk@umich.edu)

The goal of the current study is to investigate whether the acoustic properties that speakers use to encode prosodic contrasts in their production of Intonational Phrase (IP) boundaries are closely related to the properties they used to perceive those contrasts. The focus is on pause, phrase-final lengthening, and pitch reset. Thirty-two participants read sentence pairs differing in the type of boundary—word versus IP boundary. Results show that participants differed in both the combination of the acoustic properties used and the degree to which they used them to mark IP boundaries. For the perception study, the same stimuli were recorded by a model talker and the three acoustic characteristics were manipulated to systematically vary the presence and degree of the IP boundary cues in the auditory stimuli. Twenty participants from the production study participated in an eye-tracking experiment. Results of the Generalized Additive Mixed Model analyses showed limited evidence that the perception of IP boundaries was influenced by how the participants modulated the acoustic properties in production. [Work supported by NSF.]

**5pSCb5. Formant variability is actively regulated in vowel production.** Benjamin Parrell (Commun. Sci. and Disord., Univ. of Wisconsin–Madison, 1500 Highland Ave. Rm. 489, Madison, WI 53705, bparrell@wisc.edu) and Caroline A. Niziolek (Commun. Sci. and Disord., Univ. of Wisconsin–Madison, Madison, WI)

Although movement variability is typically attributed to unwanted noise in the sensorimotor system, recent work in reaching has demonstrated that variability may be at least in part actively controlled. However, it is unknown if this is true for speech production, a complex, well-practiced task controlled via non-visual sensory feedback. Here, we test how variability in formant production for vowels may be actively regulated through real-time perturbations to auditory feedback that push vowel formants toward or away from the center of each speaker's distribution for that vowel in F1/F2 space. These inward and outward perturbations effectively decrease and increase perceived variability, respectively. Participants exposed to the inward-pushing perturbation increased their produced variability both during and after exposure, indicating that lower perceived variability "frees" the motor system to be less precise. However, the outward-pushing perturbation had no consistent effect on produced variability. This suggests that at least some skilled movements are already produced at the lower limits of possible variability. While overall variability did not change, vowel "centering," a measure of within-trial correction for variability, did increase, suggesting participants became more responsive to errors. Together, these results suggest that variability in speech production is actively regulated.

**5pSCb6. Self-perception and vowel inherent spectral change.** Jonathan Jibson (Univ. of Wisconsin–Madison, 7134 Helen C White Hall, 600 N Park St., Madison, WI 53711, jibson@wisc.edu)

Speakers self-correct vowels with deviant formants: tokens whose onsets are farther from the average midpoint move toward that average, while nearer tokens move randomly (Niziolek *et al.*, 2013). The method used to establish this finding typically groups tokens based on onset formants relative to midpoint formants, and the midpoint is taken as the target. But vowel inherent spectral change research has shown that vowels are best modelled as contours rather than midpoints. The present study asks whether speakers self-correct toward trajectories rather than static midpoints. Seventeen female speakers provided 20 tokens of each stressed non-rhotic monophthong in English in [hVd] (*heed, hid,* etc.). Formants were sampled at 3% and 50% of vowel duration, and each speaker's average formant position was calculated at both samples. Sample-matched Euclidean distances were calculated—each 3% token to the 3% average, each 50% token to the 50% average—rather than comparing to the midpoint sample in both cases. Tokens were grouped based on onset distance. Far tokens showed a larger decrease in distance by the midpoint than Near tokens, and regression to the mean was ruled out as the reason. This study suggests that speakers use dynamic spectral information for online self-correction.

**5pSCb7. Stabilizing variability in the auditory feedback of speech.** Daniel Nault (Psych., Queen's Univ., 62 Arch St., Kingston, ON K7L 3N6, Canada, 14drn1@queensu.ca), David Purcell (School of Commun. Sci. and Disord., Western Univ., London, ON, Canada), and Kevin Munhall (Psych., Queen's Univ., Kingston, ON, Canada)

Auditory feedback is an essential part of speech motor control and speech learning. When feedback is perturbed in laboratory settings (e.g., Houde and Jourdan, 1998), speakers, on average, compensate for the perceived error. There is, however, considerable individual variability observed in natural speech and in speakers' responses to auditory feedback manipulations (e.g., Purcell and Munhall, 2006). Here, we introduce a novel manipulation that stabilized the predictability of auditory feedback of 20 female speakers. Participants produced the English word "head" 95 times in two different conditions. In the Control condition, subjects produced all utterances with unaltered auditory feedback. In the Stabilization condition, subjects were presented with a recording of one of their own utterances of "head" synchronized with their speech on some trials. Auditory feedback was thus made constant by playing the same recording for a set of 30 trials. Trial-to-trial variability of the talkers' speech did not change as a result of this constant feedback. Time-series analyses were performed to examine whether production variability differed among speakers in the two conditions. Results will be discussed regarding the possible role of variability in speech motor control and the importance of developing methods to detect state-change in individual time-series data.

**5pSCb8. The disruptive effect of production on learning to perceive a novel sound contrast.** Zoë Haupt (Linguist, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97405-1290, zhaupt@uoregon.edu), Tillena Trebon, Allegra Wesson, Maggie Wallace, Melissa M. Baese-Berk, Zachary Jaggers (Linguist, Univ. of Oregon, Eugene, OR), and Arthur G. Samuel (Psych., Stony Brook Univ., Stony Brook, NY)

Previous research demonstrates that simultaneous training of novel sound contrasts in both perception and production can disrupt rather than enhance perceptual learning, indicating that although perception and production are assumed to be closely connected, these modalities may have a competitive relationship. In spite of this perceptual disruption, subjects trained in perception and production show gains in producing the distinction they were trained on, compared to perception-only training. The current study examines how subjects learn to produce a new sound contrast after training in perception or production. L1 Spanish speakers were trained on an unfamiliar Basque sibilant fricative-affricate contrast: /s̺a/–/ʃa/. Since learners' productions of the contrast may not be identical to the way native

speakers distinguish it, and rather than exploring a single phonetic dimension, we apply Linear Discriminant Analysis to acoustic measurements of subjects' post-test productions to classify whether and how they distinguish the categories in a potentially multidimensional space. This classification model is then applied across conditions to compare production learning across training modes and examine how production learning relates to perceptual learning.

**5pSCb9. Establishing an index of somatosensory acuity: comparison of three measures in adults.** Olesia Gritsyk (Communicative Sci. and Disord., New York Univ., 155 E 4th St., Apt 10C, New York, NY 10009, og553@nyu.edu), Heather M. Kabakoff, Joanne Jingwen Li, Samantha Ayala (Communicative Sci. and Disord., New York Univ., New York, NY), Doug Shiller (School of Speech-Lang. Pathol. & Audiol., Univ. of Montreal, New York, NY), and Tara McAllister (Communicative Sci. and Disord., New York Univ., New York, NY)

Speech production is directed by auditory and somatosensory targets that shape and update the motor plan through corresponding feedback channels (Guenther, 2016). Somatosensory feedback is instrumental in ensuring accurate articulator placement, as shown in studies where perturbed somatosensory feedback leads to reduced speech precision (Ringel and Steer, 1963; Gammon *et al*., 1971; Jones and Munhall, 2003). As no gold standard for measuring somatosensory acuity exists, the current study will compare three measures in 20 adults: (1) an oral stereognosis task (Steele *et al*., 2014) measuring tactile input received by the articulators; (2) a novel phonetic awareness task measuring the proprioceptive sense of articulator position; (3) a bite-block task with auditory masking (Zandipour *et al*., 2006) measuring the degree of compensation for perturbation using only somatosensory feedback. To test the hypothesis that participants with higher somatosensory acuity showed larger increases in production accuracy, participants' scores on each task will be used to predict performance in an L2 vowel learning task (Li *et al*., in press). Three linear regression models will be fit, one for each somatosensory measure. The model that best explains change in production accuracy will be selected using the Akaike/Bayes Information Criteria. Establishing a valid measure of somatosensory acuity will enable future research to elucidate somatosensory influences on speech production.

**5pSCb10. Adaptation in vowels for real-time temporal perturbation.** Robin Karlin (UW-Madison, 1500 Highland Ave., Madison, WI 53705, rkarlin@wisc.edu), Chris Naber, and Benjamin Parrell (UW-Madison, Madison, WI)

Real-time altered auditory feedback has demonstrated a key role for auditory feedback in both online feedback control and in updating feedforward control for future utterances. Much of this research has examined control in the spectral domain, and has found that speakers compensate for perturbations to vowel formants, intensity, and fricative center of gravity (Houde and Jordan, 1998; Jones and Munhall 2000; Patel *et al*. 2015, inter alia). However, there is just one study that examines the regulation of temporal control via auditory feedback (Mitsuya *et al*., 2014). In the current study we introduced a real-time perturbation of speech timing to examine control of relative timing between two distinct actions (VOT for /g, k/) and inherent timing of a single action (fricative duration for /s, z/). The introduced acoustic perturbation increased the duration of the target consonant and decreased the duration of the following vowel. Overall, speakers do not compensate for lengthened consonants. However, speakers do lengthen vowel productions in response to shortening, effectively decreasing the proportion of the syllable occupied by the consonant. Vowel lengthening effects persist after perturbation has ceased. These results indicate that speakers actively monitor duration and update upcoming plans accordingly.

**5pSCb11. Increased vowel contrast induced by adaptation to a non-uniform auditory perturbation in speech.** Caroline A. Niziolek (Commun. Sci. and Disord., Univ. of Wisconsin–Madison, 1500 Highland Ave. Rm. 485, Madison, WI 53705, cniziolek@wisc.edu) and Benjamin Parrell (Commun. Sci. and Disord., Univ. of Wisconsin–Madison, Madison, WI)

When auditory feedback is perturbed in a consistent way, speakers learn to adjust their productions to compensate, a process known as sensorimotor adaptation. Typically, feedback perturbation experiments employ a transformation that targets a single vowel, or that affects all vowels in the same way, resulting in a uniform change across the vowel space (e.g., raising the first formant). Here, we examine speakers' ability to compensate for a non-uniform perturbation field which was explicitly designed to affect vowel distinctiveness, applying a shift that depended on the vowel being produced. Twenty-five participants were exposed to a "vowel centralization" feedback perturbation in which the first two formant frequencies were shifted towards the center of each participant's vowel space, making all vowels sound more like schwa. Speakers adapted to this non-uniform shift, learning to produce corner vowels with increased average vowel spacing (AVS) to partially overcome the apparent centralization. This increase in AVS remained significant after the feedback shift was removed, persisting even after a 10-minute silent period. These findings establish the validity of sensorimotor adaptation paradigms to lead to increases in vowel contrast, an outcome that has the potential to enhance intelligibility.

**5pSCb12. The role of somatosensory feedback in the production of English vowels.** Noor Al-Zanoon (Commun. Sci. and Disord., Univ. of AB, 8440 112 St. NW, Edmonton, AB T6G 2B7, Canada, alzanoon@ualberta.ca), Angela Cullum, Jacqueline Cummine (Commun. Sci. and Disord., Univ. of AB, Edmonton, AB, Canada), Caroline Jeffery (Surgery, Univ. of AB, Edmonton, AB, Canada), Bill Hodgetts, and Daniel Aalto (Commun. Sci. and Disord., Univ. of AB, Edmonton, AB, Canada)

Speech involves the complex coordination of motor, auditory and somatosensory systems. Perturbing feedback in speech has been shown to impact both vowels and consonants. The present study sought to investigate the role of somatosensory feedback in the production of four English vowels (/i, æ, u/, and /ɑ/). Thirty-three female, native English speakers were randomly assigned to control and experimental groups. The experimental group received 15 ml of 2% lidocaine mouthwash, whereas the control group received a visually comparable solution without anesthetic. Participants produced 10 repetitions of four words (/bit, bæt, but, bɑt/) in random order. Formant frequencies (F1, F2) of all four vowels were extracted and analyzed using eight linear mixed effects models. Significant effects were found for F1 in the experimental group for the single vowel /u/. The results suggest that the articulatory system is robust against somatosensory feedback perturbation in learned articulations.

# Session 5pSCc

## Speech Communication: Speech Perception in Second Language (Poster Session)

Authors will be at their posters from 2:50 p.m. to 3:35 p.m.

### *Contributed Papers*

**5pSCc1. Testing the Cue-Weighting Transfer Hypothesis with Korean listeners' perception of English lexical stress.** Hyoju Kim (Linguist, Univ. of Kansas, 1541 Lilac Ln., Blake Hall, Rm. 427, Lawrence, KS 66045-3129, kimhj@ku.edu) and Annie Tremblay (Linguist, Univ. of Kansas, Lawrence, KS)

This study investigates whether listeners' use of prosodic cues to lexical contrasts can transfer from the processing of one phonological phenomenon in the native (i.e., first) language (L1) to the processing of another phonological phenomenon in a second language (L2). It does so by investigating how Gyeongsang Korean (GK) and Seoul Korean (SK) listeners process English lexical stress contrasts. While GK has lexical pitch accents, SK does not. Korean L2 learners of English completed a sequence-recall task in which they were asked to recall four-item sequences of English words that differed suprasegmentally (stress contrast) or segmentally (phonemic contrast). The results show that there is a significant interaction between L1 and contrast type for GK listeners, but not for SK listeners. The simple effects of L1 confirm that SK listeners showed lower accuracy than GK listeners in the stress contrast condition. GK listeners' advantage over SK listeners in the stress contrast condition suggests that L2 learners can transfer the use of prosodic cues from one phonological phenomenon in the L1 (lexical pitch accents) to another phonological phenomenon in the L2 (lexical stress), providing robust evidence for the phonetic approach (e.g., Cue-Weighting Transfer Hypothesis) to the perception of lexical stress contrasts.

**5pSCc2. Perception and lexical representation of Mandarin tones by nonnative listeners.** Kuo-Chan Sun (Dept. of East Asian Studies, University of AB, Edmonton, AB T6G 2H8, Canada, kuochan@ualberta.ca)

Previous studies have shown that adult second language (L2) listeners often experience difficulty encoding language-specific phonological contrasts in word recognition. However, most research on L2 lexical representations has focused on consonants and vowels, and much remains unknown on *how* lexical tones are encoded in L2 phonological lexicon. In the current study, two experiments were conducted with twenty English learners of Mandarin and 20 Mandarin native speakers. In an ABX task, native speakers outperformed L2 listeners with higher accuracy rate and shorter response latencies. However, both groups showed poor discrimination sensitivity for pairs sharing similar tone contours (i.e., T2-T3). In a medium-lag repetition priming task, listeners were presented with a prime followed by a target that is either the same as the prime (e.g., $ni^2$-$ni^2$) or the other member of a minimal tone pair (e.g., $ni^2$-$ni^3$), eight to 20 items further down in the list. Results show that while significant facilitations in the repetition condition were observed in both groups, in the minimal-tone-pair condition (i.e., T2-T3), positive priming was observed only in the L2 group. The results of the two experiments provide insight into the interface between phonological and lexical levels in L2 spoken word recognition.

**5pSCc3. The effect of phonotactic structure on novel lexical tone perception.** Jonathan Wright (Linguist, Univ. of Oregon, Straub Hall, 1290 University of Oregon, Eugene, OR 97405, jwright8@uoregon.edu) and Melissa M. Baese-Berk (Univ. of Oregon, Eugene, OR)

Results from several studies suggest that the tonal status of the listener's first language (L1) accurately predicts novel tone discrimination ability. However, other studies show that novel tone discrimination is more complex than a simple dichotomy between tonal and non-tonal L1s, and that L1 prosodic features and acoustic properties of the target tones interact in predicting accuracy. The current study furthers this literature by examining phonotactic factors affecting novel tone discrimination. We examine the interaction between L1 and syllable structure in predicting discrimination accuracy. Tone discrimination studies typically operate under the implicit or explicit hypothesis that consonant-vowel (CV) syllables with familiar segments are the best tokens to use for tone perception tests. We test this hypothesis by including syllable structure as a predictor. This permits the examination of the interaction between L1 and syllable structure, as well as the effect of unfamiliar phonotactic structures on novel tone discrimination via consonant-consonant-vowel (CCV) syllables for native Mandarin participants and CV syllables with /ŋ/ onsets for native English participants. The results of the current study shed light on the effect of L1 background on novel tone discrimination across syllable types, as well as the effects of unfamiliar phonotactic structures on discrimination sensitivity.

**5pSCc4. Testing Korean listeners' use of acoustic cues to the /i/-/ɪ/ contrast in English spoken word recognition.** Jinmyung Lee (Linguist, Univ. of Kansas, 1541 Lilac Ln., Blake Hall 427, Lawrence, KS 66045-3129, jmlee@ku.edu) and Annie Tremblay (Linguist, Univ. of Kansas, Lawrence, KS)

Previous *offline* speech perception research has shown that English listeners rely more on spectral cues than on durational cues whereas Korean learners of English rely more on durational cues than on spectral cues when perceiving the English /i/-/ɪ/ contrast. This study uses an *online* cross-modal priming task to elucidate how L2 learners' implicit use of spectral and durational cues affects the degree of activation of words that (mis)match these cues. Auditory primes matched or mismatched the spectral cues (/i/ and /ɪ/) and durational cues (a long and short vowel duration typical of /i/ and /ɪ/, respectively) expected of visual targets (/i/ and /ɪ/). Native English listeners showed significant facilitative priming effects only when targets followed primes with matching spectral cues (primes: [gi(:)s]; target: *geese*), suggesting that they rely on spectral cues to the /i/-/ɪ/ contrast. Korean L2 learners of English at intermediate proficiency showed significant facilitative priming effects when targets followed primes with either matching or mismatching spectral and durational cues (primes: [gi(:)s], [gɪ(:)s]; target: *geese*), with the priming conditions not differing from one another. This suggests that Korean listeners did not use spectral or durational cues to the /i/-/ɪ/ contrast, possibly due to their insufficient exposure to native-like English.

**5pSCc5. The relationship between native English-speaking learners' perception and lexical representation of Hindi affricates.** Shannon Barrios (Linguist, Univ. of Utah, 255 S. Central Campus Dr., LNCO Bldg., Ste. 2300, Salt Lake City, UT 84112, s.barrios@utah.edu) and Rachel Hayes-Harb (Linguist, Univ. of Utah, Salt Lake City, UT)

Research often demonstrates discrepancies between learners' perception and encoding of novel lexical contrasts. In some cases, L2 learners appear to establish contrastive lexical representations that are undermined by online perceptual neutralization (e.g., Cutler and Weber, 2006). In others, L2 learners' target-like online perceptual representations access non-target-like lexical representations (e.g., Darcy *et al.*, 2013). Here we further investigate the

relationship between learners' perceptual and lexical representation of novel contrasts in order to better understand the source of the difficulty that they face. Two groups of naive English speakers learned 5 minimal pairs distinguished by the notoriously difficult [t͡Π]-[t͡ʃʰ] aspiration or [t͡ʃʰ]-[d͡ʒʰ] voicing contrast among Hindi affricates. Following a word learning phase, participants were tested on their perception and lexical representation of the novel contrasts. We observed that while listeners generally demonstrated perceptual sensitivity to the two affricate contrasts, they failed to encode the novel contrast lexically. Interestingly, there was a significant positive correlation between participants' perceptual acuity and their ability to contrast newly learned minimal pairs, suggesting that learner's ability to encode a novel segmental contrast may be predicted by their pre-existing perceptual sensitivity, as has been reported for lexical tone contrasts (e.g., Perrachione *et al.*, 2011).

**5pSCc6. Native English speakers' pre-existing sensitivity to the Hindi dental-retroflex contrast.** Rachel Hayes-Harb (Linguist, Univ. of Utah, 255 S Central Campus Dr., LNCO Bldg., Ste. 2300, Salt Lake City, UT 84112, r.hayes-harb@utah.edu) and Shannon Barrios (Linguist, Univ. of Utah, Salt Lake City, UT)

Native English speakers' pre-existing perceptual sensitivity to Mandarin lexical tone contrasts is associated with the ability to distinguish newly learned minimal pairs (Wong and Perrachione, 2007). Perrachione *et al.* (2011) further demonstrated that degree of pre-existing sensitivity predicts whether learners subsequently benefit more from single- or multiple-talker training. Together these findings point to the importance of pre-existing sensitivity in second language learning, yet the role of sensitivity to consonant contrasts has not been explored. We investigated variability in pre-existing sensitivity among native English speakers perceiving Hindi voiced and voiceless dental-retroflex stop contrasts. Fifty-two participants completed a web-based AXB task involving the four contrasts produced by four native Hindi speakers. Participants varied widely in their discrimination accuracy, ([t̪]-[ʈ], mean proportion correct = 0.59, min = 0.31, median = 0.59, max = .84; [t̪ʰ]-[ʈʰ], mean = 0.53, min = 0.31, median = 0.53, max = 0.75; [d̪]-[ɖ], mean = 0.56, min = 0.25, median = 0.56, max = 0.75; [d̪ʱ]-[ɖʱ], mean = 0.64, min = 0.38, median = 0.63, max = 0.81). In addition, the relative difficulty of the contrasts differed across speakers, complicating the construct of pre-existing sensitivity. We conclude that listeners exhibit variability in their pre-existing sensitivity to Hindi consonant contrasts, warranting further investigation into how this ability relates to their word learning success.

**5pSCc7. Effects of high variability phonetic training on cue reweighting in non-native vowel perception by adult Chinese learners.** Bing Cheng (Confucius Inst., Univ. of Nebraska-Lincoln, 900 N. 16th St., Nebraska Hall W205, Lincoln, NE 68588, bcheng3@unl.edu), Xiaojuan Zhang, and Dandan Qin (Lang. and Cognit. Neuroscience Lab, School of Foreign Studies, Xi'an Jiaotong Univ., Xi'an, Shaanxi, China)

While native English speakers have been found to primarily use vowel spectrum on /i/-/ɪ/ perception, Chinese learners of English dominantly use vowel duration. This study examined training effects on cue reweighting in perceiving /i/-/ɪ/ by Chinese learners. We modified the canonical HVPT paradigm with introducing variability along the secondary dimension for the contrast distinction (i.e., duration). Forty native Chinese-speaking adults were randomly assigned to two groups: the training group and the control group. Pre- and post- training tests used the tasks of natural word identification and synthetic phoneme identification. Synthesized phoneme stimuli were the two-dimensional (spectrum and duration) stimulus continua ranging from English /i/ to /ɪ/, and the natural word stimuli included both trained and untrained words produced by novel talkers. The results demonstrated that the training group showed more reliance on the primary spectral cue and less reliance on the secondary durational cue after training. The corroborating data from the word identification showed the training group improved significantly and generalized learning to new talkers and phonetic contexts. The control group did not show similar changes. These results indicated that high variability phonetic training is effective on helping L2 learners to retune attention to primary cues which facilitate generalization outcomes.

**5pSCc8. Perception of Japanese moraic-nasal (/N/) sounds followed by a vowel: A comparison of Japanese native speakers and Korean learners of Japanese.** Heesun Han (Ctr. for Int. Education and Exchange, Osaka Uviv., Intercultural Collaboration Hall, 1-1 Yamada, Suita, Osaka 565-0871, Japan, kenkyuhhs@gmail.com)

This study examines the difference in the perception of the Japanese moraic-nasal (/ɴ/) between Japanese native speakers (J) and Korean learners of Japanese (K). The intervocalic /ɴ/ is generated not solely as nasal vowels (or vowels) but includes variations such as completely closed nasal stop [ŋ] (or [ɴ]) in the oral cavity and the intermediates between these two. On the other hand, in the Korean language, it is necessary to clearly distinguish the place of articulatory (/m, n, ŋ/) on the nasal codas. Therefore, there is the possibility of differences in the perception of /ɴ/ between J and K due to differing attentiveness towards the degree of closure. This study employs the meaningful word (/goseɴeɴ/: five thousand yen) consisting of five morae. Seven Japanese native speakers are asked to naturally pronounce /goseɴeɴ/ to collect various sounds at the third mora of the test word. Sixty listeners (J:30, K:30) participated in the perception experiment and were instructed to identify stimuli by choosing one of two choices (/goseɴeɴ/, /gose:eɴ/: encouragement). The results showed that the J judgment rate of /ɴ/ is higher than that of the K. In conclusion, K depend on "the degree of closure" more than J in judgments of /ɴ/.

**5pSCc9. Perceptual discrimination measure of non-native phoneme perception in early and late Spanish-English and Japanese-English bilinguals.** Miwako Hisagi (California State Univ., Los Angeles, 5151 State University Dr., Los Angeles, CA 90032, mhisagi@hotmail.com), Eve Higby (California State Univ., East Bay, East Bay, CA), Mike Zandona (California State Univ., Los Angeles, Los Angeles, CA), Justin Kent (Occidental College, Los Angeles, CA), Daniela Castillo, Ingrid Davidovich, and Valerie Shafer (The Graduate Ctr., CUNY, New York, NY)

This study examined speech discrimination of English vowels /ɑ/ (as in "hot"), /ʌ/ (as in "hut"), and /æ/ (as in "hat") by non-native English speakers, using an AXB discrimination task. Previous research shows that a person's first language (L1) changes how speech is perceived in a second learned language (L2). Spanish and Japanese were chosen for this study because both languages share the same five spectrally different vowels, but Japanese additionally distinguishes between short and long versions of those five vowels. L2 proficiency and/or amount of L2 input were used as additional predictor variables. The target populations were early and late Spanish-English bilinguals and early and late Japanese-English bilinguals, which were compared to a control group of monolingual American English speakers. We are currently collecting data and based on previous studies (Shafer *et al.*, under review), we predict: (1) monolinguals and early-bilinguals will show better performance than late-bilinguals; (2) discrimination of /ʌ/ vs. /æ/ contrast will be easier than other contrasts, /ʌ/ vs. /ɑ/ and /ɑ/ vs. /æ/. Our findings will address how L1 spectral-temporal cues and L2 age of acquisition affect L2 speech perception.

**5pSCc10. The bilingual advantage in learning a novel accent: does specific language background modulate phonetic and phonological learning?** Mariana Vasilita (Commun., Sci. and Disord., City Univ. of New York- Brooklyn College, 2900 Bedford Ave., Brooklyn, NY 11210, mariana.vasilita34@gmail.com), Julia Wallace, and Laura Spinu (Commun. and Performing Arts, City Univ. of New York KBCC, Brooklyn, NY)

Bilinguals outperformed monolinguals in phonetic and phonological learning tasks (Tremblay and Sabourin, 2012; Antoniou *et al.*, 2015; Spinu *et al.*, 2018). Spinu *et al.* (2020) presented monolingual and bilingual participants (n = 36) with an artificial accent of English differing in four distinct ways from Standard American English: a vocalic change (diphthongization of a monophthong), consonantal change (tapping of intervocalic liquids), syllable structure change (epenthesis in s-clusters) and suprasegmental change (a novel intonation pattern in tag questions). Bilinguals outperformed monolinguals across the board but the differences were more pronounced with tapping and tag questions. Because the bilinguals' other languages were diverse (Arabic, Cantonese, Hebrew, Russian, Spanish, Urdu, Thai, and Haitian/Jamaican/St. Lucian Creole), the authors concluded that the specific languages spoken by the participants did not greatly affect the outcome, and the observed advantage correlates with the state of being bilingual. In the current study, we explore the connection between specific language background and performance with each of the four novel features. We evaluate our findings against previous claims that phonetic learning is modulated by the degree of similarity between the phonologies existent in the bilinguals' repertoire and the universal difficulty of the phonetic features learnt (Kopeckova, 2016).

**5pSCc11. The perception and production of Mandarin-accented English: The role of degree of accentedness in the interlanguage speech intelligibility benefit for talkers and the interlanguage speech intelligibility benefit for listeners.** Sheyenne Fishero (Dept. of Linguist, Univ. of Kansas, 1541 Lilac Ln., Blake Hall Rm. 427, Lawrence, KS 66045, sfishero@ku.edu), Joan Sereno, and Allard Jongman (Dept. of Linguist, Univ. of Kansas, Kansas City, KS)

The Interlanguage Speech Intelligibility Benefit for talkers (ISIB-T) claims non-native learners are better at understanding talkers with a shared L1 than they are at understanding native talkers, and the ISIB for listeners (ISIB-L) claims non-native learners are better at understanding talkers with a shared L1 than native speakers are (Hayes-Harb *et al.*, 2008). In the present study, a lexical decision task (including words with shared phonemes in both Mandarin and English, as well as words with phonemes that only occur in English) was used, with accentedness ratings of talkers and listeners collected to investigate whether ISIB is manifested differently for listeners and speakers of different proficiency levels. Accuracy scores and reaction time measures were compared for both L1 (native English) and L2 (high proficiency, low proficiency) groups when listening to strongly Mandarin-accented, weakly Mandarin-accented, and native English speech. Amount of Mandarin-accented English input was also measured. Results showed some gradience in accuracy by degree of accentedness, with trends in the predicted direction suggestive of ISIB-T and ISIB-L effects. In addition, participants were more accurate when listening to words with phonemes shared in Mandarin and English than English-specific phoneme words, an advantage that increased as the strength of speaker accent increased.

# ETHICAL PRINCIPLES OF THE ACOUSTICAL SOCIETY OF AMERICA FOR RESEARCH INVOLVING HUMAN AND NON-HUMAN ANIMALS IN RESEARCH AND PUBLISHING AND PRESENTATIONS

**The Acoustical Society of America (ASA) has endorsed the following ethical principles associated with the use of human and non-human vertebrate animals in research, and for publishing and presentations. The principles endorsed by the Society primarily follow the form of those adopted by the American Psychological Association (APA), along with excerpts borrowed or modified from the Council for International Organizations of Medical Sciences (CIOMS) and International Council for Laboratory Animal Science (ICLAS), and the American Institute of Physics Publishing (AIPP). The ASA acknowledges the difficulty in making ethical judgments, but the ASA wishes to set minimum socially accepted ethical standards for publishing in its journals and presenting at its meetings. These Ethical Principles are based on the principle that the individual author or presenter bears the responsibility for the ethical conduct of their research and its publication or presentation.**

**Authors of manuscripts submitted for publication in a journal of the ASA or presenting a paper at a meeting of the Society are obligated to follow the ethical principles of the Society. Failure to accept the ethical principles of the ASA shall result in the immediate rejection of manuscripts and/or proposals for publication or presentation. False indications of having followed the Ethical Principles of the ASA may be brought to the Ethics and Grievance Committee of the ASA.**

## I. USE OF HUMAN SUBJECTS IN RESEARCH-Applicable when human subjects are used in the research

The ASA endorses the view that all research involving human subjects requires approval by an existing appropriate governing authority (e.g., institutional review board [IRB], Health Insurance Portability and Accountability Act [HIPPA], or by other governing authorities used in many countries) whose policies are consistent with the Ethical Principles of the ASA and adopts the requirement that all research must be conducted in accordance with an approved research protocol as a precondition for participation in ASA programs. If no such governing authority exists, the research should have met the following criteria:

### Informed Consent

When obtaining informed consent from prospective participants in a research protocol, authors must have clearly and simply specified to the participants beforehand:

1. The purpose of the research, the expected duration of the study, and all procedures that were to be used.
2. The right of participants to decline to participate and to withdraw from the research in question after participation began.
3. The foreseeable consequences of declining or withdrawing from a study.
4. Anticipated factors that may have influenced a prospective participant's willingness to participate in a research project, such as potential risks, discomfort, or adverse effects.
5. All prospective research benefits.
6. The limits of confidentiality.
7. Incentives for participation.
8. Whom to contact for questions about the research and the rights of research participants. That office/person must have willingly provided an atmosphere in which prospective participants were able to ask questions and receive answers.

Authors conducting intervention research involving the use of experimental treatments must have clarified, for each prospective participant, the following issues at the outset of the research:

1. The experimental nature of the treatment;
2. The services that were or were not to be available to the control group(s), if appropriate;
3. The means by which assignment to treatment and control groups were made;
4. Available treatment alternatives if an individual did not wish to participate in the research or wished to withdraw once a study had begun; and
5. Compensation for expenses incurred as a result of participating in a study including, if appropriate, whether reimbursement from the participant or a third-party payer was sought.

### Informed Consent for Recording Voices and Images in Research

Authors must have obtained informed consent from research participants prior to recording their voices or images for data collection unless:

1. The research consisted solely of naturalistic observations in public places, and it was not anticipated that the recording would be used in a manner that could have caused personal identification or harm, or
2. The research design included deception. If deceptive tactics were a necessary component of the research design, consent for the use of recordings was obtained during the debriefing session.

### Client/Patient, Student, and Subordinate Research Participants

When authors conduct research with clients/patients, students, or subordinates as participants, they must have taken steps to protect the prospective participants from adverse consequences of declining or withdrawing from participation.

### Dispensing with Informed Consent for Research

Authors may have dispensed with the requirement to obtain informed consent when:

1. It was reasonable to assume that the research protocol in question did not create distress or harm to the participant and involves:
   a. The study of normal educational practices, curricula, or classroom management methods that were conducted in educational settings
   b. Anonymous questionnaires, naturalistic observations, or archival research for which disclosure of responses would not place participants at risk of criminal or civil liability or damage their financial standing, employability, or reputation, and confidentiality
   c. The study of factors related to job or organization effectiveness conducted in organizational settings for which there was no risk to participants' employability, and confidentiality.
2. Dispensation is permitted by law.
3. Research involving the collection or study of existing data, documents, records, pathological specimens, or diagnostic specimens, if these sources are publicly available or if the information is recorded by the investigator in such a manner that subjects cannot be identified, directly or through identifiers linked to the subjects.

### Offering Inducements for Research Participation

(a) Authors must not have made excessive or inappropriate financial or other inducements for research participation when such inducements are likely to coerce participation.

(b) When offering professional services as an inducement for research participation, authors must have clarified the nature of the services, as well as the risks, obligations, and limitations.

### Deception in Research

(a) Authors must not have conducted a study involving deception unless they had determined that the use of deceptive techniques was justified by the study's significant prospective scientific, educational, or applied value and that effective non-deceptive alternative procedures were not feasible.

(b) Authors must not have deceived prospective participants about research that is reasonably expected to cause physical pain or severe emotional distress.

(c) Authors must have explained any deception that was an integral feature of the design and conduct of an experiment to participants as early as was feasible, preferably at the conclusion of their participation, but no later than at the conclusion of the data collection period, and participants were freely permitted to withdraw their data.

### Debriefing

(a) Authors must have provided a prompt opportunity for participants to obtain appropriate information about the nature, results, and conclusions of the research project for which they were a part, and they must have taken reasonable steps to correct any misconceptions that participants may have had of which the experimenters were aware.

(b) If scientific or humane values justified delaying or withholding relevant information, authors must have taken reasonable measures to reduce the risk of harm.

(c) If authors were aware that research procedures had harmed a participant, they must have taken reasonable steps to have minimized the harm.

## II. HUMANE CARE AND USE OF NON-HUMAN VERTEBRATE ANIMALS IN RESEARCH-Applicable when non-human vertebrate animals are used in the research

The advancement of science and the development of improved means to protect the health and well-being of both human and non-human vertebrate animals often require the use of animals in research, education, and testing. The ASA remains committed to ensuring the health and welfare of vertebrate animals used for these purposes. Vertebrate animal experiments should have been undertaken only after due consideration of the relevance for health, conservation, and the advancement of scientific knowledge. (Modified from the Council for International Organizations of Medical Sciences (CIOMS) and International Council for Laboratory Animal Science (ICLAS) document: "International Guiding Principles for Biomedical Research Involving Animals-2012").

The ASA endorses the view that all research involving non-human vertebrate animals, hereinafter referred as "animals," requires approval by an existing appropriate governing authority (e.g., an institutional animal care and use committee [IACUC]) whose policies are consistent with the Ethical Principles of the ASA and adopts the requirement that all research must be conducted in accordance with an approved research protocol as a precondition for participation in ASA programs. If no such governing authority exists, the research should meet the following criteria:

1. Animals have been used only when necessary and when no alternative methods, such as non-animal approaches, mathematical models, or computer simulation, are available to achieve the scientific goals.
2. Investigators have handled all animals in compliance with all current federal, state, and local laws and regulations, and with professional standards.
3. Investigators have made all reasonable efforts to minimize the number of animals used in research to achieve the scientific goals.
4. Investigators are experienced in the care of laboratory animals, supervise all procedures involving animals, ensure all subordinates who use animals have received proper training in methodology and animal care, and assume responsibility for the comfort, health, and humane treatment of experimental animals under all circumstances.
5. The health and welfare of animals are the primary considerations in making decisions of animal care including acquisition, housing, veterinary care, and final disposition of animals.
6. All surgical procedures have been conducted under appropriate anesthesia and followed techniques that avoided infection and minimized pain during and after surgery.
7. Investigators have made all reasonable efforts to monitor and mitigate any possible adverse effects to animals as a result of the experimental protocol. Strategies to manage, mitigate, and minimize any pain and/or distress in animals should be developed in consultation with a qualified veterinarian or scientist. Animals that suffer chronic pain, distress or discomfort that cannot be relieved should be removed from the study and/or euthanized using a procedure appropriate for the species and condition of the animal.
8. Investigators proceed to rapidly and humanely terminate an animal's life when it is necessary and appropriate, always minimizing pain and always in accordance with accepted procedures as determined by a veterinarian and/or appropriate review board.

## III. PUBLICATION and PRESENTATION ETHICS-For publications in ASA journals and presentations at ASA sponsored meetings

### Statement of Ethics and Responsibilities

The mission of the ASA is to generate, disseminate, and promote the knowledge and practical applications of acoustics. To that end, it is essential that all authors of papers in ASA journals and presentors at ASA-sponsored meetings conduct themselves in accord with the highest level of professional ethics and standards.

By submitting a manuscript to an ASA journal, each author explicitly confirms that the manuscript meets the highest ethical standards. The same is required for material presented at meetings. Authors submitting to ASA journals should also adhere to the policies included in the particular journals' Instructions for Contributors.

This section is mainly based on the policies of the American Institute of Physics Publishing.

### Plagiarism

Plagiarism is the unauthorized and unacknowledged use of someone else's words, ideas, processes, data, or results in a manner that can mislead others into thinking the material is your own. Plagiarism can also be in the form of text recycling, also called self-plagiarism, where an author reuses portions of text from their own work that isn't properly credited. Plagiarism or self-plagiarism constitutes unethical scientific behavior and is never acceptable.

### Publication Credit

Authorship should be limited to those who have made a significant contribution to the concept, design, execution or interpretation of the research study. All those who have made significant contributions should be offered the opportunity to be listed as authors. The author who submits a paper for publication or an abstract for presentation and publication should ensure that all coauthors have seen the final version of the paper or abstract and have agreed to its submission. Other individuals who have contributed to the study should be acknowledged, but not identified as authors.

Proper acknowledgment of the work of others used in a research project must always be given. Information obtained privately, as in conversation, correspondence, or discussion with third parties, should not be used or reported without explicit permission from the investigator with whom the information originated. Information obtained in the course of confidential services, such as refereeing manuscripts or grant applications, cannot be used without permission of the author of the work being used.

Authors must obtain permission when reproducing or adapting any previously published materials from the original copyright holder. Proper credit lines for all previously published material must be included in the manuscript.

### Reporting Research Results

The results of research should be recorded and maintained in a form that allows analysis and review, both by collaborators before publication and by other scientists for a reasonable period after publication. Exceptions may be appropriate in certain circumstances in order to preserve privacy, to assure patent protection, or for similar reasons.

### Reporting Errors in Publication

All coauthors have an obligation to provide prompt retractions or correction of errors in published works.

### Fabrication of Data and Selective Reporting of Data

Fabrication of data is an egregious departure from the expected norms of scientific conduct, as is the selective reporting of data with the intent to mislead or deceive, as well as the theft of data or research results from others.

### Disclosure of Conflicts of Interest

A conflict of interest is anything that interferes with, or could reasonably be perceived as interfering with, the full and objective presentation of articles in the ASA journals and presentations at the ASA meetings. Author(s) have the obligation to disclose any personal interest or relationship that has the potential to be affected by publication of the submitted manuscript or presentation at ASA meeting:

1. The complete affiliation(s) of each author and sources of funding for the published or presented research should be clearly described in the paper or publication abstract.
2. If the publication or presentation of the research would directly lead to the financial gain of the author(s), then a statement to this effect must appear in the acknowledgment section of the paper or presentation abstract or

in a footnote of a paper. Authors must report any financial interest in corporate or commercial entities dealing with the subject matter of the manuscript or presentation.

3. If the research that is to be published or presented is in a controversial area and the publication or presentation presents only one view in regard to the controversy, then the existence of the controversy and this view must be provided in the acknowledgment section of the paper or presentation abstract or in a footnote of a paper. It is the responsibility of the author to determine if the paper or presentation is in a controversial area and if the person is expressing a singular view regarding the controversy.

Authors must submit corrections if conflicts of interests are revealed after publication.

Approved by the Executive Council on ??/??/20??.

# Sustaining Members of the Acoustical Society of America

The Acoustical Society is grateful for the financial assistance being given by the Sustaining Members listed below and invites applications for sustaining membership from other individuals or corporations who are interested in the welfare of the Society.

Application for membership may be made to the Executive Director of the Society and is subject to the approval of the Executive Council. Dues of $1000.00 for small businesses (annual gross below $100 million) and $2000.00 for large businesses (annual gross above $100 million or staff of commensurate size) include a subscription to the *Journal* as well as a yearly membership certificate suitable for framing. Small businesses may choose not to receive a subscription to the *Journal* at reduced dues of $500/year.

Additional information and application forms may be obtained from Elaine Moran, Office Manager, Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300. Telephone: (516) 576-2360; E-mail: elaine@acousticalsociety.org

### Acentech Incorporated
*www.acentech.com*
Cambridge, Massachusetts
Consultants in Acoustics, Audiovisual and Vibration

### ACO Pacific Inc.
*www.acopacific.com*
Belmont, California
Measurement Microphones, the ACOustic Interface™ System

### Acoustics First Corporation
*www.acousticsfirst.com*
Richmond, Virginia
Materials to Control Sound and Eliminate Noise™

### American Institute of Physics
*www.aip.org*
College Park, Maryland
Career resources, undergraduate education, science policy, and history

### BBN Technologies
*www.bbn.com*
Cambridge, Massachusetts
R&D company providing custom advanced research based solutions

### GRAS Sound and Vibration
*www.gras.us*
Twinsburg, Ohio
Measurement microphones, intensity probes, calibrators

### Kinetics Noise Control, Inc.
*www.kineticsnoise.com*
Dublin, Ohio
Kinetics manufactures products to address vibration and noise control, room acoustics, and seismic restraint concerns for almost any building application

### Massa Products Corporation
*www.massa.com*
Hingham, Massachusetts
Design and Manufacture of Sonar and Ultrasonic Transducers
Computer-Controlled OEM Systems

### Meyer Sound Laboratories, Inc.
*www.meyersound.com*
Berkeley, California
Manufacture Loudspeakers and Acoustical Test Equipment

### National Council of Acoustical Consultants
*www.ncac.com*
Indianapolis, Indiana
An Association of Independent Firms Consulting in Acoustics

### National Gypsum Company
*www.nationalgypsum.com*
Charlotte, North Carolina
Manufacturer of acoustically enhanced gypsum board

### Raytheon Company
**Integrated Defense Systems**
*www.raytheon.com*
Portsmouth, Rhode Island
Sonar Systems and Oceanographic Instrumentation: R&D
in Underwater Sound Propagation and Signal Processing

### ROXUL, Inc. – Core Solutions (OEM)
*www.roxul.com*
Milton, ON, Canada
Offers a variety of insulation products ranging in density and dimension to meet any production requirements. Products are successfully used in numerous acoustical OEM applications providing solutions for a number of industries

### Thales Underwater Systems
*www.thales-naval.com*
Somerset, United Kingdom
Prime contract management, customer support services, sonar design and production, masts and communications systems design and production

### 3M Personal Safety Division (PSD)
*www.3m.com/occsafety*
Minneapolis, Minnesota
Products for personal and environmental safety, featuring E·A·R and Peltor brand hearing protection and fit testing, Quest measurement instrumentation, audiological devices, materials for control of noise, vibration, and mechanical energy, and the E·A·RCALSM laboratory for research, development, and education, NVLAP-accredited since 1992.
**Hearing conservation resource center**
*www.e-a-r.com/hearingconservation*

### Wenger Corporation
*www.wengercorp.com*
Owatonna, Minnesota
Design and Manufacturing of Architectural
Acoustical Products including Absorbers, Diffusers, Modular Sound Isolating Practice Rooms, Acoustical Shells and Clouds for Music Rehearsal and Performance Spaces

### Wyle Laboratories
*www.wyle.com*
Arlington, Virginia
The Wyle Acoustics Group provides a wide range of professional services focused on acoustics, vibration, and their allied technologies, including services to the aviation industry

ACOUSTICAL · SOCIETY · OF · AMERICA

# APPLICATION FOR SUSTAINING MEMBERSHIP

The Bylaws provide that any person, corporation, or organization contributing annual dues as fixed by the Executive Council shall be eligible for election to Sustaining Membership in the Society.

Dues have been fixed by the Executive Council as follows: $1000 for small businesses (annual gross below $100 million); $2000 for large businesses (annual gross above $100 million or staff of commensurate size). Dues include one year subscription to *The Journal of the Acoustical Society of America* and programs of Meetings of the Society. Please do not send dues with application. Small businesses may choose not to receive a subscription to the *Journal* at reduced dues of $500/year. If elected, you will be billed.

Name of Company ______________________________

Address ______________________________

Telephone: ______________ Fax: ______________

E-mail: ______________ WWW: ______________

Size of Business: ☐ Small business ☐ Small business—No Journal ☐ Large business

Type of Business ______________________________

**Please enclose a copy of your organization's brochure.**

In listing of Sustaining Members in the Journal and on the ASA homepage we should like to indicate our products or services as follows:

______________________________

(please do not exceed fifty characters)

Name of company representative to whom journal should be sent:

______________________________

It is understood that a Sustaining Member will not use the membership for promotional purposes.

Signature of company representatives making application:

______________________________

Please send completed applications to: Executive Director, Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300, (516) 576-2360, asa@acousticalsociety.org

## MEMBERSHIP INFORMATION AND APPLICATION INSTRUCTIONS

Applicants may apply for one of four grades of membership, depending on their qualifications: Student Member, Associate Member, Corresponding Electronic Associate Member or full Member. To apply for Student Membership, fill out Parts I and II of the application; to apply for Associate, Corresponding Electronic Associate, or full Membership, or to transfer to these grades, fill out Parts I and III.

| **BENEFITS OF MEMBERSHIP** | full Member | Associate | ce-Associate | Student |
|---|---|---|---|---|
| *JASA* Online–Vol. 1 (1929) to present | * | * | * | * |
| *JASA* tables of contents e-mail alerts | * | * | * | * |
| *JASA*, printed | * | * | | |
| *JASA Express Letters*–online | * | * | * | * |
| *Acoustics Today*–the quarterly magazine | * | * | * | * |
| Proceedings of Meetings on Acoustics | * | * | * | * |
| *Noise Control* and *Sound*, *It's Uses and Control*–online archival magazines | * | * | * | * |
| *Acoustics Research Letters Online* (ARLO)–online archive | * | * | * | * |
| Programs for Meetings | Online | Online | Online | Online |
| Meeting Calls for Papers | Online | Online | Online | Online |
| Reduced Meeting Registration Fees | * | * | | * |
| Society Membership Directory | Online | Online | Online | Online |
| Electronic Announcements | * | * | * | * |
| *Physics Today* | * | * | * | * |
| Eligibility to vote and hold office in ASA | * | | | |
| Eligibility to be elected Fellow | * | * | | |
| Participation in ASA Committees | * | * | * | * |

## QUALIFICATIONS FOR EACH GRADE OF MEMBERSHIP AND ANNUAL DUES

**Student:** Any student interested in acoustics who is enrolled in an accredited college or university for half time or more (at least eight semester hours). Dues: $50 per year.
**Associate:** Any individual interested in acoustics. Dues: $115 per year. After five years, the dues of an Associate increase to that of a full Member.
**Corresponding Electronic Associate:** Any individual residing in a developing country who wishes to have access to ASA's online publications only including *The Journal of the Acoustical Society of America* and Meeting Programs [see http://acousticalsociety.org/membership/membership_and_benefits]. Dues $50 per year.
**Member:** Any person active in acoustics, who has an academic degree in acoustics or in a closely related field or who has had the equivalent of an academic degree in scientific or professional experience in acoustics, shall be eligible for election to Membership in the Society. A nonmember applying for full Membership will automatically be made an interim Associate Member, and must submit $115 with the application for the first year's dues. Election to full Membership may require six months or more for processing; dues as a full Member will be billed for subsequent years.

## JOURNAL OPTIONS AND COSTS FOR FULL MEMBERS AND ASSOCIATE MEMBERS ONLY

- ONLINE JOURNAL. All members will receive access to the *The Journal of the Acoustical Society of America (JASA)* at no charge in addition to dues.
- PRINT JOURNAL. Twelve monthly issues of *The Journal of the Acoustical Society of America*. **Cost: $35 in addition to dues.**
- **EFFECTIVE DATE OF MEMBERSHIP**. If your application for membership and dues payment are received by 15 September, your membership and Journal subscription will begin during the current year and you will receive all back issues for the year. If you select the print journal option. If your application is received after 15 September, however, your dues payment will be applied to the following year and your Journal subscription will begin the following year.

## OVERSEAS AIR DELIVERY OF JOURNALS

Members outside North, South, and Central America can choose to have print journals sent by air freight at a cost of $185 in addition to dues. JASA on CD-ROM is sent by air mail at no charge in addition to dues.

# ACOUSTICAL SOCIETY OF AMERICA

1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300, asa@acousticalsociety.org

| For Office Use Only |
|---|
| Dues Rcvd ____________ |
| Aprvd by Ed ____________ |
| Aprvd by EC____________ |

## APPLICATION FOR MEMBERSHIP

Applicants may apply for one of four grades of membership, depending on their qualifications: Student Member, Associate Member, Corresponding Electronic Associate Member or full Member. To apply for Student Membership, fill out Parts I and II of this form; to apply for Associate, Corresponding Electronic Associate, or full Membership, or to transfer to these grades, fill out Parts I and III.

### PART I. TO BE COMPLETED BY ALL APPLICANTS (Please print or type all entries)

CHECK ONE BOX IN EACH COLUMN ON THE RIGHT

☐ NON-MEMBER APPLYING FOR:
☐ MEMBER REQUESTING TRANSFER TO:

☐ STUDENT MEMBERSHIP
☐ ASSOCIATE MEMBERSHIP
☐ CORRESPONDING ELECTRONIC ASSOCIATE MEMBERSHIP
☐ FULL MEMBERSHIP

Note that your choice of journal option *may* increase or decrease the amount you must remit.

### SELECT JOURNAL OPTION:

**Student members** will automatically receive access to The Journal of the Acoustical Society of America online at no charge in addition to dues. Remit $45.

**Corresponding Electronic Associate Members** will automatically receive access to The Journal of the Acoustical Society of America and Meeting Programs online at no charge in addition to dues. Remit $50.

Applicants for **Associate or full Membership** must select one Journal option from those listed below. Note that your selection of journal option determines the amount you must remit.

[ ] Online access only—$150
[ ] Online access plus print Journal $185

Applications received after 15 September: Membership and Journal subscriptions begin the following year.

**OPTIONAL AIR DELIVERY:** Applicants from outside North, South, and Central America may choose air freight delivery of print journals for an additional charge of $185. If you wish to receive journals by air, remit the additional amount owed with your dues. JASA on CD-ROM is sent by air mail at no charge in addition to dues.

| | | | |
|---|---|---|---|
| LAST NAME | FIRST NAME | MIDDLE INITIAL | MS/MR/MRS/DR/PROF |
| HOME ADDRESS (STREET & NUMBER) | | | |
| CITY | STATE OR PROVINCE | ZIP OR POSTAL CODE | COUNTRY |
| NAME OF ORGANIZATION OR BUSINESS | | | |
| DEPARTMENT | | | |
| ORGANIZATION ADDRESS (STREET & NUMBER) | | | |
| CITY | STATE OR PROVINCE | ZIP OR POSTAL CODE | COUNTRY |
| BUSINESS TELEPHONE: AREA CODE/NUMBER | FAX: AREA CODE/NUMBER | HOME TELEPHONE: AREA CODE/NUMBER | |
| E-MAIL ADDRESS: (PRINT CLEARLY) | MOBILE PHONE: AREA CODE/NUMBER | | |
| DATE AND PLACE OF BIRTH (Req'd for Awards and Emeritus Status) | SEX: ☐ Female ☐ Male ☐ Non-Binary ☐ Transgender ☐ Prefer not to answer ☐ | | |
| HIGHEST ACADEMIC DEGREE | DATE OF DEGREE | FIELD | INSTITUTION GRANTING DEGREE |
| OTHER DEGREE | MONTH/YEAR | FIELD | INSTITUTION GRANTING DEGREE |

CHECK PERFERRED ADDRESS FOR MAIL: ☐ HOME ☐ ORGANIZATION

**Part I Continued ➛**

**PART I CONTINUED:** ACOUSTICAL AREAS OF INTEREST TO APPLICANT. Indicate your three main areas of interest below, using 1 for your main interest, 2 for your second, and 3 for your third interest. (DO NOT USE CHECK MARKS.)

- ☐ ACOUSTICAL OCEANOGRAPHY **M**
- ☐ ANIMAL BIOACOUSTICS **L**
- ☐ ARCHITECTURAL ACOUSTICS **A**
- ☐ BIOMEDICAL ACOUSTICS **K**
- ☐ COMPUTATIONAL ACOUSTICS **O**
- ☐ ENGINEERING ACOUSTICS **B**
- ☐ MUSICAL ACOUSTICS **C**
- ☐ NOISE & NOISE CONTROL **D**
- ☐ PHYSICAL ACOUSTICS **E**
- ☐ PSYCHOLOGICAL & PHYSIOLOGICAL ACOUSTICS **F**
- ☐ SIGNAL PROCESSING IN ACOUSTICS **N**
- ☐ SPEECH COMMUNICATION **H**
- ☐ STRUCTURAL ACOUSTICS & VIBRATION **G**
- ☐ UNDERWATER ACOUSTICS **J**

## PART II: APPLICATION FOR STUDENT MEMBERSHIP

| | | |
|---|---|---|
| NAME AND ADDRESS OF COLLEGE OR UNIVERSITY WHERE PRESENTLY ENROLLED | | |
| DEGREE EXPECTED | MONTH & YEAR DEGREE EXPECTED | NUMBER OF SEMESTER HOURS ATTENDED THIS SEMESTER |
| PRINT NAMES & E-MAIL ADDRESSES OF TWO FACULTY MEMBERS CERTIFYING THAT YOU ARE REGISTERED FOR AT LEAST ONE-HALF OF FULL TIME | | |
| SIGNATURES OF THE TWO FACULTY MEMBERS LISTED ABOVE CERTIFYING THAT YOU ARE REGISTERED AT LEAST HALF TIME | | |
| SIGNATURE OF APPLICANT | | DATE |

## PART III: APPLICATION FOR ASSOCIATE MEMBERSHIP, CORRESPONDING ELECTRONIC ASSOCIATE MEMBERSHIP OR FULL MEMBERSHIP (and interim Associate Membership)

SUMMARIZE YOUR MAJOR PROFESSIONAL EXPERIENCE on the lines below: list employers, duties and position titles, and dates, beginning with your present position. Attach additional sheets if more space is required.

| |
|---|
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| |
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| |
| CONTRIBUTIONS TO ACOUSTICS: LIST MAIN PUBLICATIONS, PATENTS, ETC. Attach separate sheets if required. |
| |
| |
| |

SPONSORS AND REFERENCES: An application for full Membership requires the names, and email addresses of two references who must be **full Members or Fellows** of the Acoustical Society. Names and signatures are NOT required for Associate Membership, Corresponding Electronic Associate Membership or Student Membership applications.

| | |
|---|---|
| PRINT NAME OF REFERENCE (required for Full Member applications only) | PRINT NAME OF SECOND REFERENCE (required for Full Member applications only) |
| EMAIL ADDRESS OF REFERENCE | EMAIL ADDRESS OF SECOND REFERENCE |
| SIGNATURE OF APPLICANT | DATE |

**MAIL THIS COMPLETED APPLICATION, WITH APPROPRIATE PAYMENT TO:** ACOUSTICAL SOCIETY OF AMERICA, 1305 WALT WHITMAN ROAD, SUITE 300, MELVILLE, NY 11747-4300; FAX: 631-923-2875

## METHOD OF PAYMENT

☐ Check or money order enclosed for $ ____________ (U.S. funds/drawn on U.S. bank)

☐ American Express ☐ VISA ☐ MasterCard Signature ______________________________________________

*(Credit card orders must be signed)*

Account Number ☐☐☐☐☐☐☐☐☐☐☐☐☐☐☐☐

Expiration Date Mo. ☐☐ Yr. ☐☐

Security Code ☐☐☐☐

Due to security risks and Payment Card Industry (PCI) data security standards e-mail is NOT an acceptable way to transmit credit card information. Please return this form by Fax (631-923-2875) or by postal mail.

# Regional Chapters and Student Chapters

Anyone interested in becoming a member of a regional chapter or in learning if a meeting of the chapter will be held while he/she is in the local area of the chapter, either permanently or on travel, is welcome to contact the appropriate chapter representative. Contact information is listed below for each chapter representative.

Anyone interested in organizing a regional chapter in an area not covered by any of the chapters below is invited to contact the Cochairs of the Committee on Regional Chapters for information and assistance: Evelyn Hoglund, Ohio State University, hoglund1@osu.edu and Kenneth W. Good, Jr., Armstrong World Industries, Inc., kwgoodjr@armstrong.com

### AUSTIN STUDENT CHAPTER
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www.acoustics.byu.edu

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### CHICAGO
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celmer@hartford.edu

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### LOS ANGELES
Neil A. Shaw
www.asala.org

### MICHIGAN STUDENT CHAPTER
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asdougl@umich.edu

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dbacoustics@cox.net

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### PENNSYLVANIA STATE UNIVERSITY STUDENT CHAPTER
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mtn5048@psu.eduwww.psuasa.org

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