IHCON 2018

International Hearing Aid Research Conference

August 15 – 19, 2018

Granlibakken Conference Center
Tahoe City, California
IHCON 2018

Sponsors

● -- ● -- ●

National Institute on Deafness
and Other Communication Disorders

The Hearing Industry Research Consortium

Massachusetts Eye and Ear
Eaton-Peabody Laboratories
IHCON 2018

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Birger Kollmeier

Technical Co-Chairs

Virginia Best  •  Tom Francart

Organizational Co-Chairs

Sunil Puria  •  Sigfrid Soli

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National Acoustic Laboratories, Australia

Shilpi Banerjee
Pacific University, USA

Carol Mackersie
San Diego State University, USA

Conference Coordinator: Barbara Serrano  •  Conference Support: Kevin O’Connor
# Student Scholarship Recipients

<table>
<thead>
<tr>
<th>Name</th>
<th>Affiliation</th>
<th>Country</th>
</tr>
</thead>
<tbody>
<tr>
<td>Braden Carei</td>
<td>Belmont University</td>
<td>USA</td>
</tr>
<tr>
<td>Florian Denk</td>
<td>University of Oldenburg</td>
<td>Germany</td>
</tr>
<tr>
<td>Benjamin Dieudonné</td>
<td>KU Leuven</td>
<td>Belgium</td>
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<tr>
<td>Anastasia Grindle</td>
<td>Northern Illinois University</td>
<td>USA</td>
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<tr>
<td>Jose Guerreiro</td>
<td>University of Strathclyde</td>
<td>United Kingdom</td>
</tr>
<tr>
<td>Hao Lu</td>
<td>University of Minnesota</td>
<td>USA</td>
</tr>
<tr>
<td>Rémi Marchand</td>
<td>The HEARing CRC</td>
<td>Australia</td>
</tr>
<tr>
<td>Kristi Oeding</td>
<td>University of Minnesota</td>
<td>USA</td>
</tr>
<tr>
<td>Chhayakant Patro</td>
<td>University of Minnesota</td>
<td>USA</td>
</tr>
<tr>
<td>Trevor Perry</td>
<td>University of Minnesota</td>
<td>USA</td>
</tr>
<tr>
<td>Varsha Rallapalli</td>
<td>Northwestern University</td>
<td>USA</td>
</tr>
<tr>
<td>Ilja Reinten</td>
<td>Academic Medical Center</td>
<td>Netherlands</td>
</tr>
<tr>
<td>Raul Sanchez-Lopez</td>
<td>Technical University of Denmark</td>
<td>Denmark</td>
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<tr>
<td>Jonathan Vaisberg</td>
<td>Western University</td>
<td>Canada</td>
</tr>
<tr>
<td>Tilde Van Hirtum</td>
<td>KU Leuven</td>
<td>Belgium</td>
</tr>
<tr>
<td>Yashuo Wu</td>
<td>University of Illinois</td>
<td>USA</td>
</tr>
<tr>
<td>Britt Yazel</td>
<td>University of California, Davis</td>
<td>USA</td>
</tr>
<tr>
<td>Time</td>
<td>Wednesday, August 15</td>
<td>Thursday, August 16</td>
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<tr>
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</tr>
<tr>
<td>7:00 AM</td>
<td>Breakfast</td>
<td>Breakfast</td>
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<td>7:30 AM</td>
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<tr>
<td>8:00 AM</td>
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<tr>
<td>9:00 AM</td>
<td>Poster Session I (Paired to B1, B2, B3)</td>
<td>Poster Session II (Paired to C3, C4, C5)</td>
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<tr>
<td>9:30 AM</td>
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<tr>
<td>11:00 AM</td>
<td>B2. Objective Measures for Assessing Aided and Unaided Hearing; Multimodal</td>
<td>C4. Models and their Application to Hearing Aids</td>
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<tr>
<td>11:30 AM</td>
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<td>12:00 PM</td>
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<td>12:30 PM</td>
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<tr>
<td>1:00 PM</td>
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<td>2:00 PM</td>
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<tr>
<td>7:00 PM</td>
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<tr>
<td>7:30 PM</td>
<td>A1. Opening Remarks and Keynote Address</td>
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<tr>
<td>8:00 PM</td>
<td>Discussion</td>
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<tr>
<td>8:30 PM</td>
<td>Social</td>
<td>Social and Posters</td>
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<tr>
<td>9:00 PM</td>
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</table>

* Candidates for the 2020 IHCON Technical Committee (TC)
## Program Summary

### Session A1
**Chair:** Birger Kollmeier  
**Session Name/Title:** A1. Opening Remarks and Keynote Address

<table>
<thead>
<tr>
<th>Date and Time</th>
<th>Session ID</th>
<th>Chair/Presenting Author</th>
<th>Session Name/Title</th>
</tr>
</thead>
<tbody>
<tr>
<td>Wed. Aug. 15</td>
<td>A1-O-0</td>
<td>Sunil Puria, Sengfrid Soli, Birger Kollmeier</td>
<td>Opening Remarks</td>
</tr>
<tr>
<td>19:30-19:45</td>
<td>A1-O-1</td>
<td>Nima Mesgarani</td>
<td>Brain-controlled assisted hearing technologies: challenges and opportunities</td>
</tr>
<tr>
<td>19:45-20:30</td>
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</table>

### Session B1
**Chair:** Judy Dubno  
**Session Name/Title:** B1. Physiologically Steered Hearing Devices

<table>
<thead>
<tr>
<th>Date and Time</th>
<th>Session ID</th>
<th>Chair/Presenting Author</th>
<th>Session Name/Title</th>
</tr>
</thead>
<tbody>
<tr>
<td>Thu. Aug. 16</td>
<td>B1-O-1</td>
<td>Volker Hohmann</td>
<td>Decoding auditory attention from eye gaze</td>
</tr>
<tr>
<td>08:00-08:40</td>
<td>B1-O-2</td>
<td>Graham Naylor</td>
<td>On the role of head movements and beam width when listening through a gaze-controlled acoustic beamformer</td>
</tr>
<tr>
<td>08:40-09:20</td>
<td>B1-O-3</td>
<td>Simon Dodd</td>
<td>Cognitive-Driven Binaural Speech Enhancement System for Hearing Aid Applications</td>
</tr>
<tr>
<td>09:00-09:40</td>
<td>B1-O-4</td>
<td>Tao Zhang</td>
<td>A Joint Auditory Attention Decoding and Adaptive Binaural Beamforming Algorithm for Hearing Devices</td>
</tr>
<tr>
<td>11:10-11:30</td>
<td>B1-P-01</td>
<td>Saren A. Fuglsang</td>
<td>Cortical EEG entrainment to attended speech in hearing-impaired listeners</td>
</tr>
<tr>
<td>11:30-11:50</td>
<td>B1-P-02</td>
<td>Alejandro Lopez Valdes</td>
<td>Waxy business: the impact of ear cerumen on in-ear electrophysiological recordings.</td>
</tr>
<tr>
<td>11:50-12:10</td>
<td>B1-P-03</td>
<td>Jonathan Marcher-Rastred</td>
<td>Closed-loop BCI control of acoustic feedback using selective auditory attention</td>
</tr>
<tr>
<td>12:10-12:30</td>
<td>B1-P-04</td>
<td>Jayagannesh Swaminathan</td>
<td>Electroencephalography based encoding and decoding of speech in hearing impaired listeners</td>
</tr>
<tr>
<td>12:30-12:50</td>
<td>B1-P-05</td>
<td>Tao Zhang</td>
<td>An EEG Database for a Multi-Talker, Noisy and Reverberant Environment for Hearing Device Applications</td>
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</table>

### Session B2
**Chair:** Suzanne Levy  
**Session Name/Title:** B2. Objective Measures for Assessing Aided and Unaided Hearing; Multimodal

<table>
<thead>
<tr>
<th>Date and Time</th>
<th>Session ID</th>
<th>Chair/Presenting Author</th>
<th>Session Name/Title</th>
</tr>
</thead>
<tbody>
<tr>
<td>Thu. Aug. 16</td>
<td>B2-O-1</td>
<td>Elke Verschuieren</td>
<td>Objectively and automatically measuring speech intelligibility: the effect of contextual information in the stimulus</td>
</tr>
<tr>
<td>11:10-11:30</td>
<td>B2-O-2</td>
<td>Britt Yazel*</td>
<td>Tracking the dynamics of selective attention and listening fatigue in a noisy conversation with EEG, eye tracking, and pupillometry</td>
</tr>
<tr>
<td>11:30-11:50</td>
<td>B2-O-3</td>
<td>Lars Bramslaw</td>
<td>Segregation benefit from deep neural networks assessed via speech recognition and physiological measures</td>
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<tr>
<td>11:50-12:10</td>
<td>B2-P-01</td>
<td>Robin Gransier</td>
<td>Intersubject variability of phase-locked activity to a wide range of modulation frequencies assessed with auditory steady-state responses</td>
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<tr>
<td>12:10-12:30</td>
<td>B2-P-02</td>
<td>Petteri Hyvärinen</td>
<td>Peripheral auditory-evoked responses obtained with the Ear-EEG</td>
</tr>
<tr>
<td>12:30-12:50</td>
<td>B2-P-03</td>
<td>Karolina Kluk</td>
<td>Speech-Auditory Brainstem Responses in Individuals with Hearing Loss; Effects of Aiding and Background Noise</td>
</tr>
<tr>
<td>12:50-13:10</td>
<td>B2-P-04</td>
<td>Karolina Kluk</td>
<td>Effects of noise exposure on young adults with normal audiometric hearing</td>
</tr>
<tr>
<td>13:10-13:30</td>
<td>B2-P-05</td>
<td>Hye Jin Lim</td>
<td>Is it enough to use the objective hearing test for pediatric hearing rehabilitation? The importance of behavioral pure tone audiometry through auditory training with hearing aids</td>
</tr>
<tr>
<td>13:50-14:10</td>
<td>B2-P-07</td>
<td>Chhayakant Patro*</td>
<td>Individual Differences in the Strength of Middle-Ear Muscle Reflex in Young Adults with Normal Hearing and Varying Degree of Noise Exposure</td>
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<tr>
<td>14:10-14:30</td>
<td>B2-P-08</td>
<td>Jonathan Pietrobon</td>
<td>Application of real time recurrent neural networks for estimating probe tube insertion depth</td>
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<tr>
<td>14:50-15:10</td>
<td>B2-P-10</td>
<td>Patrick Johannes Schäfer</td>
<td>Stimulus reconstruction as a tool to decode the orientation of spatial auditory attention in a four speaker free field environment</td>
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<tr>
<td>15:10-15:30</td>
<td>B2-P-11</td>
<td>Patrick Johannes Schäfer</td>
<td>A Granger causality based method to decode the orientation of spatial auditory attention exclusively from the EEG</td>
</tr>
<tr>
<td>15:30-15:50</td>
<td>B2-P-12</td>
<td>Hye Yoon Seol</td>
<td>Impact of hearing aid noise reduction algorithms on speech-evoked auditory brainstem response</td>
</tr>
<tr>
<td>15:50-16:10</td>
<td>B2-P-13</td>
<td>Larissa Taylor</td>
<td>Listening effort and speech intelligibility measurement for multiple subjects simultaneously in complex listening scenarios</td>
</tr>
<tr>
<td>16:10-16:30</td>
<td>B2-P-14</td>
<td>Jana Van Canney*</td>
<td>Effect of envelope shape on the amplitude of the following response</td>
</tr>
<tr>
<td>16:30-16:50</td>
<td>B2-P-15</td>
<td>Dorothea Wendt</td>
<td>Impact of SNR, hearing impairment, and hearing aid technology on listening effort as indicated by the pupillary response</td>
</tr>
<tr>
<td>16:50-17:10</td>
<td>B2-P-16</td>
<td>William M. Whitemer</td>
<td>The difference between objective and subjective speech intelligibility</td>
</tr>
<tr>
<td>17:10-17:30</td>
<td>B2-P-17</td>
<td>Yashuo Wu*</td>
<td>Study of abnormal Phone Perception in Children with Reading Disability</td>
</tr>
</tbody>
</table>

### Session B3
**Chair:** Stefan Launer  
**Session Name/Title:** B3. Individualized Diagnostics, Fitting, and Rehabilitation: Part I

<table>
<thead>
<tr>
<th>Date and Time</th>
<th>Session ID</th>
<th>Chair/Presenting Author</th>
<th>Session Name/Title</th>
</tr>
</thead>
<tbody>
<tr>
<td>Thu. Aug. 16</td>
<td>B3-O-1</td>
<td>Susan Scollie</td>
<td>Hearing aid fitting and the individual: integrating measures of dynamic range and audibility with individualized diagnostics</td>
</tr>
<tr>
<td>17:00-17:40</td>
<td>B3-O-2</td>
<td>Tobias Neher</td>
<td>Hearing aid processing strategies for listeners with different auditory profiles: Insights from the BEAR project</td>
</tr>
<tr>
<td>17:40-18:00</td>
<td>B3-O-3</td>
<td>David Eddins</td>
<td>Novel hearing instrument and fitting protocol for treating hyperacusis</td>
</tr>
<tr>
<td>18:00-18:20</td>
<td>B3-P-01</td>
<td>Jörg-Hendrik Bach</td>
<td>Explaining fine-tuning gain differences for subjects with similar audiograms</td>
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<tr>
<td>18:20-18:40</td>
<td>B3-P-02</td>
<td>Arthur Boothroyd</td>
<td>Application of open-source speech processing platforms to independent hearing-aid research: two examples.</td>
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<tr>
<td>18:40-19:00</td>
<td>B3-P-03</td>
<td>Benjamin Casswell-Midwinter</td>
<td>Discrimination of Frequency-Gain Curve Adjustments</td>
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<tr>
<td>19:00-19:20</td>
<td>B3-P-04</td>
<td>Raul Sanchez-Lopez*</td>
<td>Data-driven auditory profiling as a tool for defining Better HeAring Rehabilitation (BEAR)</td>
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<tr>
<td>19:20-19:40</td>
<td>B3-P-05</td>
<td>Blair Ellis</td>
<td>Psychoacoustic factors influencing speech perception in noise and reverberation for individuals with and without hearing impairment</td>
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<tr>
<td>19:40-19:50</td>
<td>B3-P-06</td>
<td>Michal Fereczkowski</td>
<td>Psychoacoustic factors in cochlear compression in the clinic</td>
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<tr>
<td>19:50-20:00</td>
<td>B3-P-07</td>
<td>Paula Folkard</td>
<td>Fit-to-target and SII normative data for DSL v5.0 adult fittings</td>
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<tr>
<td>20:00-20:10</td>
<td>B3-P-08</td>
<td>Anastasia Grindel*</td>
<td>Improving predictive real-ear-to-coupler difference predictions and aided audibility using immittance</td>
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<tr>
<td>20:10-20:20</td>
<td>B3-P-09</td>
<td>Meng Guo</td>
<td>An Evaluation Method for Simultaneous Assessment of Ability to Match Target Gain, Sound Quality, and Feedback Performance in Hearing Aids</td>
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<tr>
<td>20:20-20:30</td>
<td>B3-P-10</td>
<td>Sijan Guo</td>
<td>Masking Effects on Perceptual Cues in Hearing-Impaired Ears</td>
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<tr>
<td>20:30-20:40</td>
<td>B3-P-11</td>
<td>Benjamin Hornsby</td>
<td>Using the Vanderbilt Fatigue Scale to explore the effects of hearing loss and device use on listening-related fatigue in adults and children with hearing loss</td>
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<tr>
<td>20:40-20:50</td>
<td>B3-P-12</td>
<td>C. T. Justine Hui</td>
<td>Comparison of unaided and aided psychoacoustic tuning curves with normal hearing and hearing impaired listeners</td>
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<tr>
<td>20:50-21:00</td>
<td>B3-P-13</td>
<td>Homayoun Kamkar-Parsi</td>
<td>Towards individualized automatic control of directionality strength in binaural hearing aids</td>
</tr>
<tr>
<td>21:00-21:10</td>
<td>B3-P-14</td>
<td>Robert Koch</td>
<td>Skills Transference Validity of a Probe Tube Placement Training Simulator</td>
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<tr>
<td>21:10-21:20</td>
<td>B3-P-15</td>
<td>Sören Laugesen</td>
<td>Towards a clinically viable spectro-temporal modulation test</td>
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<tr>
<td>21:20-21:30</td>
<td>B3-P-16</td>
<td>Suzanne Levy</td>
<td>Real-world achieved gains and subjective outcomes of a wide-bandwidth contact hearing aid fitted with CAM2</td>
</tr>
<tr>
<td>Thu. Aug. 16</td>
<td>Session BX</td>
<td>Chair: Sunil Puria</td>
<td>BX. Lessons Learned and Future IHCON Developments</td>
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<tr>
<td>19:50-20:40</td>
<td>BX-O-1</td>
<td>Sigfrid Soli</td>
<td>How did we get here? A brief history of IHCON</td>
</tr>
<tr>
<td>20:15-20:40</td>
<td>BX-O-2</td>
<td>Birger Kollmeier, Sunil Puria, Virginia Best, Tom Francart, Sigfrid Soli</td>
<td>Podium Discussion</td>
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<thead>
<tr>
<th>Fri. Aug. 17</th>
<th>Session C3</th>
<th>Chair: Peggy Nelson</th>
<th>C3. Individualized Diagnostics, Fitting, and Rehabilitation: Part II</th>
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<tbody>
<tr>
<td>08:00-08:20</td>
<td>C3-O-6</td>
<td>Brian Moore</td>
<td>Assessing the Fidelity of Envelope Coding: The Envelope Regularity Discrimination (ERD) Test</td>
</tr>
<tr>
<td>08:20-08:40</td>
<td>C3-O-5</td>
<td>Ilja Reiniten*</td>
<td>Subjective evaluation of different attack times in single microphone noise reduction</td>
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<tr>
<td>08:40-09:00</td>
<td>C3-O-6</td>
<td>Jonathan Vaisberg*</td>
<td>Preferred hearing aid gain settings for music-listening using a 3D modified simplex procedure implemented with the Open Source Master Hearing Aid platform</td>
</tr>
<tr>
<td>09:00-09:40</td>
<td>C3-O-7</td>
<td>Christophe Michéy</td>
<td>Perceptual strategies for consonant discrimination in individuals with and without hearing loss</td>
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<tr>
<td>09:20-09:40</td>
<td>C3-O-8</td>
<td>Christopher Slugocki</td>
<td>Development of an integrated Repeat and Recall Test (RRT)</td>
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<tr>
<td>11:50-12:10</td>
<td>C3-P-17</td>
<td>Carol Mackersie</td>
<td>Hearing-aid self-adjustment by experienced hearing-aid users using the Goldilocks protocol</td>
</tr>
<tr>
<td>12:10-12:30</td>
<td>C3-P-18</td>
<td>Rémi Marchand*</td>
<td>Factors influencing enjoyment of music with hearing aids</td>
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<tr>
<td>12:30-12:50</td>
<td>C3-P-19</td>
<td>Ryan McCreery</td>
<td>Comparisons of criteria for prescriptive target proximity in children who wear hearing aids</td>
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<tr>
<td>12:50-13:10</td>
<td>C3-P-20</td>
<td>James Miller</td>
<td>Aided Sentence Perception as Related to the Identification of Syllable Constituents and the Use of Context</td>
</tr>
<tr>
<td>13:10-13:30</td>
<td>C3-P-21</td>
<td>Elaine Ng</td>
<td>Hearing aid experience and background noise affect the robust relationship between working memory and speech reception in noise</td>
</tr>
<tr>
<td>13:30-13:50</td>
<td>C3-P-22</td>
<td>Kristi Geding*</td>
<td>The effect of carrier bandwidth and intensity on amplified spectral modulation detection thresholds</td>
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<tr>
<td>13:50-14:10</td>
<td>C3-P-23</td>
<td>Jon Øygarden</td>
<td>An app for validation of hearing aid fitting with speech and noise presented in three sound channels</td>
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<tr>
<td>14:10-14:30</td>
<td>C3-P-24</td>
<td>Trevor Perry*</td>
<td>Comparing self-adjusted amplification and gain preferences across listening contexts</td>
</tr>
<tr>
<td>14:30-14:50</td>
<td>C3-P-25</td>
<td>Patrick Reidman*</td>
<td>The Effect of Hearing Aid Amplification on Emotion Recognition and Emotional Responses</td>
</tr>
<tr>
<td>14:50-15:10</td>
<td>C3-P-26</td>
<td>Karrie Recker</td>
<td>10 Years of Acceptable Noise Level Test, What Have We Learned?</td>
</tr>
<tr>
<td>15:10-15:30</td>
<td>C3-P-27</td>
<td>Benjamin Casewell-Midwinter</td>
<td>Subjective Judgments on Frequency-Gain Curve Adjustments</td>
</tr>
<tr>
<td>15:30-15:50</td>
<td>C3-P-28</td>
<td>Saul Sanchez-Lopez</td>
<td>Auditory tests for characterizing individual hearing deficits: The BEAR test battery</td>
</tr>
<tr>
<td>15:50-16:10</td>
<td>C3-P-29</td>
<td>Josef Schlitterlacher</td>
<td>Estimation of auditory filter shapes across frequencies using machine learning</td>
</tr>
<tr>
<td>16:10-16:30</td>
<td>C3-P-30</td>
<td>Michael Schulte</td>
<td>Evaluation of noise reduction systems with subjective listening effort measurements</td>
</tr>
<tr>
<td>16:30-16:50</td>
<td>C3-P-31</td>
<td>Gurjit Singh</td>
<td>A pre-post intervention study of hearing aid amplification: Results of the Emotional Communication in Hearing Questionnaire (EMO-ChEQ)</td>
</tr>
<tr>
<td>16:50-17:10</td>
<td>C3-P-32</td>
<td>Maaike Van Eckhoutte</td>
<td>The effect of extended bandwidth on loudness and DSL v5.0 targets</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
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<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>11:10-11:50</td>
<td>C4-O-1</td>
<td>Torsten Dau</td>
<td>Auditory processing models and their potential application in hearing technology</td>
</tr>
<tr>
<td>11:50-12:10</td>
<td>C4-O-2</td>
<td>Jan Bruce</td>
<td>Overcoming the Quagmire of Hearing Loss Heterogeneity: Toward Optimization of Hearing Aids via Distinct Genetic Hearing Loss Populations</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Fri. Aug. 17</th>
<th>Session C5</th>
<th>Chair: Anna Diedesch</th>
<th>C5. Binaural Hearing Aids and Fitting</th>
</tr>
</thead>
<tbody>
<tr>
<td>17:00-17:30</td>
<td>C5-O-1</td>
<td>Virginia Best</td>
<td>Spatial hearing, hearing loss and hearing aids</td>
</tr>
<tr>
<td>17:30-17:50</td>
<td>C5-O-2</td>
<td>Benjamin Dieudonné</td>
<td>A semantic framework for binaural mechanisms in speech understanding</td>
</tr>
<tr>
<td>17:50-18:10</td>
<td>C5-O-3</td>
<td>Michelle Molis</td>
<td>Effects of Hearing Loss and Broad Binaural Fusion on Dichotic Concurrent Vowel Identification</td>
</tr>
<tr>
<td>18:10-18:30</td>
<td>C5-O-4</td>
<td>Florian Denk*</td>
<td>Can we trust our aided ears? Acoustic directional information available in different hearing device styles</td>
</tr>
<tr>
<td>18:30-18:50</td>
<td>C5-O-5</td>
<td>Jorge Mejía</td>
<td>Intelligibility, subjective preference and cortical measures for listeners using a binaural beamformer in realistic background noise at high signal to noise ratio</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Sat. Aug. 18</th>
<th>Session D6</th>
<th>Chair: Chas Pavlovic</th>
<th>D6. Signal Processing for Hearing Aids</th>
</tr>
</thead>
<tbody>
<tr>
<td>08:00-08:20</td>
<td>D6-O-1</td>
<td>James Kates</td>
<td>Combining Remote Microphones with Hearing Aid Processing</td>
</tr>
<tr>
<td>08:20-08:40</td>
<td>D6-O-2</td>
<td>DeLiang Wang</td>
<td>A deep learning based segregation algorithm to improve speech intelligibility of hearing-impaired listeners in reverberant-noisy conditions</td>
</tr>
<tr>
<td>08:40-09:00</td>
<td>D6-O-3</td>
<td>Marc Aubreville</td>
<td>Deep Neural Networks for Noise Reduction under Hearing Aid Side Conditions</td>
</tr>
<tr>
<td>09:00-09:20</td>
<td>D6-O-4</td>
<td>Birger Kollmeier</td>
<td>Accessible infrastructure for hearing research: A commodity-hardware-based mobile prototype of a hearing aid featuring the openMHA.org research software platform</td>
</tr>
<tr>
<td>09:20-09:40</td>
<td>D6-P-1</td>
<td>Lars Bramesew</td>
<td>Influence of signal enhancement algorithms on auditory movement detection in acoustically complex situations</td>
</tr>
<tr>
<td>09:40-09:50</td>
<td>D6-P-2</td>
<td>Benjamin Dieudonné*</td>
<td>Head shadow enhancement to improve sound localization and speech intelligibility in noise</td>
</tr>
<tr>
<td>09:50-10:10</td>
<td>D6-P-3</td>
<td>Simon Duclo</td>
<td>Comparison of binaural MVDR-based beamforming algorithms using an external microphone</td>
</tr>
<tr>
<td>10:10-10:30</td>
<td>D6-P-4</td>
<td>Jose Guerreiro*</td>
<td>Fully-Adaptive Embedded MEMS Acoustic Sensor System</td>
</tr>
<tr>
<td>10:30-10:50</td>
<td>D6-P-5</td>
<td>Chiho Harada</td>
<td>Prototype of Deep Neural Network-based Real-Time Speech Enhancement System and Its Evaluation</td>
</tr>
<tr>
<td>10:50-11:00</td>
<td>D6-P-6</td>
<td>Kostas Kakkouais</td>
<td>An algorithm for reverberation suppression in cochlear implants</td>
</tr>
<tr>
<td>11:00-11:20</td>
<td>D6-P-7</td>
<td>Petri Korhonen</td>
<td>Modulation Spectrum as an Additional Quantifier of Hearing Aid Processing</td>
</tr>
</tbody>
</table>

IHCON 2018
### Paired Posters

**D6-P-06** Borys Kowalewski  
Effects of hearing-aid amplification on consonant audibility and forward masking

**D6-P-09** Tobias Neher  
Influence of individual factors and acclimatization on the perception of noise reduction settings

**D6-P-10** Kohto Nozaki  
System implementation of dereverberation method using exponential averaging with attack and release time constants for hearing aids

**D6-P-11** Varsha Rallapalli  
Effects of multichannel compression on spectral contrast of vowels processed by real hearing aids

**D6-P-12** Daniel Rasethwane  
Electroacoustic and Behavioral Evaluation of an Open Source Audio Processing Platform

**D6-P-13** Marina Salorio-Corbetto  
The number of amplitude-compression channels and compression speed on the intelligibility of speech in noise

**D6-P-14** Gokce Sarar  
Sparsity promoting adaptive beamforming for hearing aids

**D6-P-15** Henning Scheckter  
Evaluation of acoustic feedback cancellation for a multi-microphone earpiece using a null-steering beamformer

**D6-P-16** Masahiro Sunchana  
Prototype of Four-Channel Low-Latency Real-Time Blind Source Separation System for Hearing Aids

**D6-P-17** Tine Van Hirtum  
The potential of speech envelope enhancement for auditory intervention in dyslexia

**D6-P-18** Alan Winberg  
Compensation for impaired temporal processing

### Session D7

**Chair:** Kathryn Arehart

**Sat. Aug. 18**

| 11:00-11:40 | D7-O-1 Andrew Sabin | Trends influencing hearing devices and real-world efficacy of a self-fit method |
| 11:40-12:00 | D7-O-2 Eric Hoover | Blinded comparison of premium hearing aids and personal sound amplification products |
| 12:00-12:20 | D7-O-3 Nicolas Bisgaard | Modeling hearing aid coverage in different countries and estimating value of treatment |

**D7-P-01** William Audette  
An Open Source Platform for Testing Hearing Aid Algorithms

**D7-P-02** Young-Soo Chang  
Predicting the cochlear dead regions in patients with hearing loss through a machine learning based approach: A preliminary study

**D7-P-03** Young Sang Cho  
A Comparison of Performance on Speech Intelligibility in Noise between Personal Sound Amplification Products (PSAPs), Basic Hearing Aids, and Premium Hearing Aids

**D7-P-04** Drew Dundas  
The Wider Bandwidth of the Earlens Contact Hearing Aid Leads to Superior Sound Quality for Streamed Audio

**D7-P-05** Chiho Hanuta  
DNN-based fitting formula for hearing aids – the minimum quantity of data for sufficient training

**D7-P-06** Nicola Hildebrand  
Hearing Aid Users’ Interactions with a Smart Phone Fine Tuning App

**D7-P-07** Matthias Latzel  
Do people with (very) mild hearing loss benefit from modern hearing instruments?

**D7-P-08** Vinaya Manchahia  
Benefits and shortcomings of Direct-to-Consumer Hearing Devices (OCHD): Analysis of large secondary data from Amazon user reviews using mixed method approach

**D7-P-09** Graham Naylor  
Early results from a study of clinical data routinely collected across the VA Audiology service: patterns and pitfalls in extracting meaning from 731,209 hearing aid fittings

**D7-P-10** Suyeon Park  
Investigating the effect of hearing aid wireless connectivity technology on speech intelligibility and listening effort

**D7-P-11** Niels Henrik Pontoppidan  
EVOPTION Big data supporting public hearing health policies

**D7-P-12** S. R. Prakash  
A sequential approach to Trainable Hearing Aids

**D7-P-13** Kimberly Skinner  
Hearing Aid Maintenance by Older Adults: Time Course and Intervention

**D7-P-14** Matthew Waggenspack  
Evaluating and Improving Accessibility of Over-the-Counter Hearing Aid Devices

### Session D8

**Chair:** Karolina Kluk

**Sat. Aug. 18**

| 17:00-17:40 | D8-O-1 Joerg Buchholz | Bringing the real world into the lab and vice versa: More realistic assessment of hearing ability and device benefit |
| 17:40-18:00 | D8-O-2 Laurent Simon | Perceptual comparison of 3D audio reproduction system with hearing devices: a focus on localization |
| 18:00-18:20 | D8-O-3 Inga Holube | What’s going on? Individualized evaluation in the real world |
| 18:20-18:40 | D8-O-4 Karolina Smeds | Paired comparisons of preference for hearing-aid settings measured in the field and in the laboratory – Evaluation of LEAP, a new laboratory test based on the CoSS framework |
| 18:40-19:00 | D8-O-5 Annelies Devesse | The effect of age on real-life listening |

**D8-P-01** Dragana Barac-Cikoja  
Sound Advice: An Interactive Learning Environment for Optimizing Use of Hearing Assistive Technology

**D8-P-02** Donald Hayes  
Benchmarking detection and classification in automatic hearing aids

**D8-P-03** Lorienne Jenstad  
Ecological Momentary Assessment: A Field Evaluation of Subjective Ratings of Speech in Noise as a Function of Acoustic and Task Variables

**D8-P-04** Erik Jorgensen  
Temporal Stability of Auditory Ecology of Adult Hearing Aid Users

**D8-P-05** Ulfik Kowalk  
An open source toolkit for privacy-preserving real-world EMA data collection

**D8-P-06** Hao Lu  
Tracking eye and head movements in natural conversational settings: Effects of age, hearing loss and background noise level

**D8-P-07** Markus Meis  
Research design and pre-post evaluation of hearing aid provision under field and laboratory controlled conditions: Contributions for Postmarket Surveillance requirements

**D8-P-08** Andrea Micula  
Effect of task difficulty and signal processing on recall performance and recall strategy

**D8-P-09** Bhavisha Parmar  
The spatial speech test for assessing binaural hearing

**D8-P-10** Erin Picou  
The effects of microphone technology on word recognition and behavioral listening effort in school-aged children

**D8-P-11** Hasan Saleh  
An Update of the Connected Speech Test

**D8-P-12** Braden Care  
Real-time Simulations of Hearing Loss and Auditory Prostheses

**D8-P-13** Petra von Gablenz  
Data analysis from real-world hearing assessment

### Explanation of the Session IDs

A–D indicates the day of the conference (A=Wednesday, B=Thursday, C=Friday, D=Saturday);  
1–8 or X identifies the session topic;  
O or P indicates an oral or poster presentation; and  
1, 2, … or 01, 02, … uniquely specifies each talk or poster within a given topic. 

**Bold** = invited speaker  
*( * = scholarship recipient)*
SESSION A1. Keynote Address

Session Chair: Birger Kollmeier

A1-O-1: Brain-controlled assistive hearing technologies: Challenges and opportunities

Nima Mesgarani
Columbia University, USA

Listening in noisy and crowded environments is exceptionally challenging for hearing-impaired listeners. Assistive hearing devices can suppress certain types of background noise, but they cannot help a user attend to a single conversation amongst many without knowing which speaker is attended. Recent advances in scientific discoveries of speech processing in the human auditory cortex have motivated several new paths to enhance the efficacy of hearable technologies. These possibilities include speech neuroprosthesis which aims to establish a direct communication channel with the brain, auditory attention decoding where the similarity of a listener’s brainwave to the sources in the acoustic scene is used to identify the attended source, and increased speech perception using electrical brain stimulation. In parallel, the field of speech signal processing has recently seen tremendous progress due to the emergence of deep learning models, where even solving the cocktail party problem is no longer out of reach. I will discuss the recent efforts in bringing together the latest progress in brain-computer interfaces and speech processing technologies to design and actualize the next generation of assistive hearing devices, with the potential to augment speech communication in realistic and challenging acoustic conditions.
Novel multi-microphone binaural algorithms for hearing devices show significantly higher spatial selectivity than established algorithms, such as adaptive directional microphones. Furthermore, these algorithms have scene analysis capabilities that allow to estimate direction, activity and even the type of the sound sources surrounding the hearing aid user. This will allow, in principle, to provide the user with enhanced signals for improved acoustic communication. It is unclear, however, how to select and present the sound source or sound sources that are in the focus of attention of the listener. Several approaches have been proposed to estimate this “hearing wish” using bio-signal inputs, including EEG and head and eye movement sensor signals, but they have not yet been shown to be fast and accurate enough to steer the hearing aid and to provide better acoustic communication in realistic conditions than established technology. To further investigate the combination of hearing-wish estimation with speech enhancement algorithms, this contribution presents data and simulations on using head and eye-movement sensor data for hearing wish estimation and algorithm control.

Speech comprehension in a coordinated response measure task with four simultaneous spatially-separated speakers was measured in 14 normal hearing listeners in an audiovisual interactive environment. The task of the subject was to report the color and number uttered by the speaker who said the keyword “Goethe”, i.e., the subject had to steer spatial attention to the direction of that target speaker. Results show that responses were never correct when the subject indicated the direction of target speaker incorrectly, supporting the view that spatial attention is a major factor in decoding complex scenes. While performing the task, the subject’s gaze direction was recorded by electrooculography (EOG) combined with a head tracking system, which would be feasible also in hearing aids. No instructions were given to the subjects regarding head and eye movements during the task. A realtime-capable algorithm was developed that estimated the spatial attention of the subject in each trial of the speech test from the observed head- and eye movements in combination with an acoustic analysis of the spatio-temporal distribution of source positions. This estimate was then compared with the true position of the target speaker in each trial. Results show that the target source can be estimated from the gaze direction with about 85% hit rate across all trials of all subjects. An application in a simulated spatially selective algorithm demonstrated a significant SNR benefit based on the individual gaze estimate. This shows that eye gaze maybe a relevant factor in estimating the hearing wish, and should be investigated further in other realistic conditions. Realtime implementation in an open tool, the open Master Hearing Aid (openMHA) will be done in future work. [Work funded by DFG FOR1732, SFB/TR31, EXC 1077, EU MSCA GA 675324 and NIH R01DC015429. Research reported in this presentation was supported by the NIDCD of the NIH. The content is solely the responsibility of the authors and does not necessarily represent the official views of the NIH.]
B1-O-2: On the role of head movements and beam width when listening through a gaze-controlled acoustic beamformer

Graham Naylor¹, Thomas Lunner², Bernd Porr³, and Lubos Hladek⁴

¹ University of Nottingham
² Eriksholm Research Centre, Oticon A/S, Denmark
³ University of Glasgow

Under certain circumstances, directional microphones steered by eye-gaze might provide greater speech-in-noise benefits than conventional head-steered systems, because it is natural to look at the person one is listening to, and people’s head orientation tends to undershoot the lateral angle of target speakers. However, any eye-steering benefit might be limited by beamformer directivity, time required to re-orient gaze to a new target, and the degree of undershoot in the individual listener’s head orientations.

This study investigated speech perception when listening through simulated directional microphones to a target with dynamically changing location in spatially distributed background noise. Infrared tracking of the head and eyes provided real-time control of signal levels to each of a ring of loudspeakers. For experimental contrasts, the orientation of the virtual beamformer was controlled either by horizontal head+eye-gaze angle (GAZE) or head angle (HEAD), and two beam widths provided overall noise attenuation of 8dB (WIDE) or 12dB (NARROW) with respect to a simulation of omnidirectional microphones (OMNI). SNR was adjusted per participant (n=18) and per condition to achieve equal intelligibility for beamformer lobes locked to the target. Every 1.5 seconds, the participants heard a single target word coming from one of two possible locations, 30° left or right. Target location changed without warning, with visual indication of target location occurring simultaneously with acoustical signal onset.

When the target location remained static, speech intelligibility was better with GAZE than with HEAD control, but only when the beam was NARROW. The benefit of GAZE over HEAD control was greatest for those participants who moved their heads least from the midline. When targets changed location without warning, performance dropped substantially. In this case, the WIDE HEAD condition showed the best average performance, because it provided the best chance to hear out a target from the opposite direction. Furthermore, the benefit of HEAD over GAZE control was greatest for those participants who moved their heads least from the midline.

This study grossly simplified several aspects of conversational interaction, and adjusted SNR to equate reference performance across conditions. Hence the wider validity of the detailed results might be quite limited. Nevertheless, the results suggest there is no straightforward answer to the question of whether eye-steered directional microphones would be beneficial. It is likely to depend on an individual’s propensity to head movement, as well as the specific physical arrangement of talkers, strength of microphone directivity, and vital pre-emptive markers of conversational turn-taking.

B1-O-3: Cognitive-driven binaural speech enhancement system for hearing aid applications

Ali Aroudi¹, Daniel Marquardt², and Simon Doclo¹

¹ Dept. of Medical Physics and Acoustics and Cluster of Excellence Hearing4All, University of Oldenburg
² Starkey Hearing Technologies

During the last decades significant progress has been made in multi-microphone speech enhancement algorithms for hearing aids. Nevertheless, the performance of most algorithms depends on correctly identifying the target speaker to be enhanced. To identify the target speaker from single-trial EEG recordings in an acoustic scenario with two competing speakers, a least-squares-based auditory attention decoding (AAD) method was proposed in [1]. While the performance of this
method has been mainly studied for noiseless and anechoic conditions, it is important to fully understand its performance in realistic noisy and reverberant conditions. In addition, the considered AAD method typically assumes that the clean speech signals of both the attended and the unattended speaker are available as reference signals for decoding, while in practice obviously only the microphone signals at the ears, containing reverberation, background noise and interference, are available.

In this contribution, we investigate AAD for different acoustic conditions (anechoic, reverberant, noisy, reverberant-noisy) [2]. In particular, we investigate the impact of different acoustic conditions for AAD filter training and decoding using trials of 30 sec. Furthermore, aiming at enhancing the target speaker in a noisy and reverberant environment with two competing speakers, we propose a cognitive-driven speech enhancement system, consisting of a direction-of-arrival estimator, steerable binaural beamformers and AAD. For an acoustic scenario with two competing speakers and diffuse babble noise, 64-channel EEG responses with 18 participants were recorded. First, we show that for all considered acoustic conditions it is possible to decode auditory attention, even when the acoustic conditions for AAD filter training and decoding are different. Second, when using reference signals affected by reverberation and/or background noise, a comparable decoding performance as when using clean reference signals can be obtained. In contrast, when using reference signals affected by the interfering speaker, the decoding performance significantly decreases. Third, we evaluate the performance of the proposed cognitive-driven speech enhancement system in terms of the intelligibility-weighted signal-to-interference-plus-noise ratio averaged over all (correctly as well as wrongly decoded) trials and all participants.

References:


vex optimization approach. The proposed algorithm has two advantages over the existing algorithms. First, the optimization objective aims to balance auditory attention alignment, target speech distortion, noise and interference suppression. Secondly, there is no need to estimate the speech envelope of each talker from the noisy and reverberant mixture which is a very challenging problem in practice. The proposed algorithm was evaluated using a newly recorded EEG database for a multi-talker, noisy and reverberant environment (Xiao et al, 2018). The evaluation results confirm the effectiveness of the proposed algorithm.

POSTER SESSION I

Paired to Sessions B1, B2, and B3

Thursday 9:40 AM – 11:10 AM,
8:40 PM – 10:00 PM

Posters for Session I should be put up by 8:00 AM Thursday, August 16, and taken down after 10:00 PM Thursday, August 16. Presenters should be at their posters from 9:40 AM – 11:10 AM

B1-P-01: Cortical EEG entrainment to attended speech in hearing-impaired listeners
Søren A. Fuglsang¹, Jonatan Märcher-Rørsted¹, Torsten Dau¹,², and Jens Hjortkjær¹
¹ Technical University of Denmark
² Hearing Systems Group

Robust single-trial electroencephalogram (EEG) measures of auditory attention in normal-hearing (NH) listeners suggest a possible role of EEG in attention-controlled brain computer interfaces (BCI). However, to date it remains unclear whether the attentional focus of hearing-impaired (HI) listeners can be reliably decoded from single-trial EEG responses to speech mixtures. Here, we investigated the influence of a symmetric sensorineural hearing impairment on the EEG correlates of selective auditory attention. We recorded EEG responses to speech in quiet and in the presence of a competing talker in 22 older HI listeners and 22 age-matched NH controls. In the competing talker condition with two simultaneous speakers of opposite sex, the two speech streams were loudness matched and the gender of the to-be-attended speaker was randomized. The speech stimuli were amplified to provide equal audibility to the HI and NH listeners. The listeners answered comprehension questions related to the content of the attended speech. Similar high accuracies in both the single-talker and the competing talker conditions were found for both listener groups. However, when listening selectively to speech masked by competing speech, HI listeners rated attentive listening to be significantly more difficult than the NH controls. Cortical EEG entrainment to envelopes of attended speech was found to be stable against competing speech in both listener groups. The stable cortical EEG synchronization to attended speech enabled single-trial decoding of auditory attention with equally high classification accuracies in HI and NH listeners. The results thus suggest a robust cortical representation of attended speech in HI listeners at neutral (~0 dB) target-to-masker intensity ratios and are promising for real-time BCI hearing aid applications.

B1-P-02: Waxy business: The impact of ear cerumen on in-ear electrophysiological recordings
Lotte Simone Emilie Petersen¹,², Alejandro Lopez Valdes³, Renskje K. Hietkamp², Mike Lind Rank, and Thomas Lunnør¹,⁴
¹ University of Southern Denmark
² Eriksølm Research Centre, Oticon A/S
³ UNEEG Medical A/S
⁴ Linköping University

Recent research, investigating bioelectrical signals recorded from within the ear, has demonstrated that it is feasible to record auditory evoked potentials such as the Auditory Steady State Response by placing recording electrodes inside the ear cavity (Mikkelsen KB et al. 2015; Fiedler L et al. 2016; Christensen CB et al. 2018).
The current lab-based in-ear recording methodology considers extensive ear cleaning as well as skin and electrode preparation to favour the quality of the electrophysiological signals. This consideration together with the use of contact enhancing electrolyte gels or paste is incompatible with realistic long-term recordings. One promising solution to this problem is to employ dry, metal-oxide electrodes that are not dependent on electrolyte media (Searle A and Kirkup L, 2000; Kappel S et al. 2018). However, little remains known about the impact of an uncleared or unprepared ear on the quality of in-ear recordings. The outer ear produces cerumen that consists of shed skin cells, hair, and the secretions of the glands of the ear canal, thus, introducing an extra layer between the dry electrodes and the skin.

We hypothesised that the presence of cerumen in the ear cavity would have an impact on the quality of auditory evoked potentials recorded from the ear. We investigated this by measuring Auditory Steady State Responses from 8 participants recorded before and after cleaning and preparing the ears and with a replicate 3 weeks later to allow for cerumen recovery. The amount of cerumen present in the ear was evaluated visually with an otoscopic camera before both recording sessions and rated by 3 clinical experts.

Two-tailed paired t-tests showed no significant effect of cleaning of the ears nor of the presence of cerumen in the ear cavity after correcting for multiple comparisons. A Spearman’s rank correlation showed no significant correlation between the amount of cerumen present in the ear and the amplitude of the auditory steady state responses.

Given the small cohort of participants in this study, caution should be exercised when interpreting these results. However, there seems to be no evidence to indicate that cleaning of the ears or ear preparation should be a mandatory step when recording auditory evoked potentials inside the ear when using dry, metal-oxide electrodes. Additional participants should be recruited to achieve statistical power in our observations.

Recent work showing single-trial EEG decoding of selective auditory attention has perspectives for ‘cognitively steered’ hearing instruments that selectively amplify attended sound streams. Yet, previous work has relied on offline EEG measures of auditory attention and little is known about the dynamics of real-time attention-steered feedback. Here, we present results of a closed-loop system in which real-time decoding of attention was used to steer acoustic feedback of two competing talkers. The system first extracts the audio envelope of each talker and synchronizes the audio streams with the band-pass filtered EEG. Real-time classification of the attended talker is based on canonical correlations between the EEG and the audio envelopes fed to a linear classifier. The classification output is then used to control the relative gain of each speech stream. We present results from closed-loop experiments examining the influence of neurofeedback on listeners’ ability to maintain and switch attention between two spatially separated competing speech streams. By presenting simultaneous speech streams, each with embedded auditory 1-back targets, we monitor behavioral performance and BCI system performance throughout the progression of a trial. Moreover, we use questionnaires to evaluate the perceived experience with closed-loop BCI control when listening to naturalistic speech mixtures. The presented data suggests that attention switching may be possible even when the previously ignored stream is substantially suppressed.

B1-P-03: Closed-loop BCI control of acoustic feedback using selective auditory attention

Jonatan Märcher-Rørsted¹, Soren A. Fuglsang¹, Daniel E. Wong², Torsten Dau³, Alain de Cheveigné², and Jens Hjorthjær¹

¹ Technical University of Denmark
² École normale supérieure
³ Hearing Systems Group

B1-P-04: Electroencephalography based encoding and decoding of speech in hearing impaired listeners

Yaqing Su, Narayan Sankaran, and Jayaganesh Swamianthan

Starkey Hearing Research Center

There has been a recent surge in interest in developing hearing aids that incorporate electroencephalography (EEG) signals as an input to guide signal processing to facilitate robust speech perception in hearing-impaired (HI) listeners. Although such an approach seems enticing, the vast majority of research in this topic has been conducted in listeners with normal-hearing. Given the peripheral degradation with hearing loss and the concomitant effects such degradation may impose on the coding of speech features, fundamental questions remain on even the feasibility of extracting meaningful EEG information from HI listeners that may be used to inform HA signal processing algorithms. The overall goal of this study was to explore the feasibility of using EEG to study the effects
of hearing loss on the speech processing deficits in HI listeners.

Rather than using continuous speech, HI listeners were presented a set of nonsense vowel-consonant-vowel (VCV) speech. This choice was primarily made to simplify the interpretation of the EEG responses. We measured both perceptual and EEG responses to a closed set of VCV stimuli in listeners with moderate hearing loss. Responses were measured for unamplified and amplified conditions. A state-of-the-art machine-learning classifier was trained to discriminate the EEG signal evoked by each consonant. The performance of the classifier was compared to the HI listeners’ psychophysical performance.

For all HI listeners, performance in the perceptual discrimination task was poor even with sound amplification. EEG waveforms from the HI listeners showed different patterns of responses for each consonant. A simple classifier could ‘decode’ consonants from the EEG signal. Under certain conditions, a relationship between the neural and perceptual representation of consonants could be established in HI listeners. Implications for technologies aiming to use neural signals to guide hearing aid processing will be discussed.

**B1-P-05: An EEG database for a multi-talker, noisy and reverberant environment for hearing device applications**

Jinjun Xiao¹, Sina Miran², Behtash Babadi², and Tao Zhang¹

¹ Starkey Hearing Technologies

Most of the existing electroencephalography (EEG) databases on auditory attention are based on audio stimuli with two talkers and without much noise or reverberation. This work reports a recently-recorded EEG database based on audio stimuli representing a multi-talker, noisy and reverberant environment. A set of binaural audio stimuli were generated using a set of head-related transfer functions (HRTFs) in a simulated noisy and reverberant room. The listener sat at the center of the room with talkers sitting around the listener. The background babble noise was generated using twenty-four loudspeakers distributed in the room. The talkers are all set to the same level and the babble noise is 5dB lower than each talker. The resulting audio stimuli were presented to the listener using a set of ER-3 insert earphones at a “loud but comfortable” level in a sound-treated and semi-electrically shielded sound booth. The listener was instructed to perform a binaural listening task and attend to one of the talkers at a time. A total of 12 normal hearing listeners were used. Each subject listened to a total of 50 minutes of stories with each being split into multiple 1-minute segments. A 64-channel scalp EEG system was used to record the listener’s response. In this paper, we will present the detailed experiment design and a preliminary analysis of the collected data. In addition, we will demonstrate the usefulness of this database using a near real-time auditory attention decoding algorithm (Miran et al, 2018).

### Thursday, August 16

**SESSION B2. Objective Measures for Assessing Aided and Unaided Hearing: Multimodal**

Session Chair: Suzanne Levy

**B2-O-1: Objectively and automatically measuring speech intelligibility: The effect of contextual information in the stimulus**

Eline Verschueren, Jonas Vanthornhout, and Tom Francart

KU Leuven - University of Leuven
In current clinical practice, hearing aid fitting mainly relies on the patient’s subjective response to tones. The use of non-speech stimuli is suboptimal because understanding speech, and not hearing tones, is the main goal of wearing hearing aids. In addition, the necessary active participation of the patient can make these sessions challenging. Therefore, we aim to develop an objective measure of speech intelligibility, based on brain responses, to provide the basis for individualized and self-fitting hearing aids and neuro-steered devices. As we want our measure to assess how well the hearing aid processes speech, we want to minimize the effects of patient-related linguistic skills. In this study we investigated the influence of top-down semantic processing on our objective measure.

We recorded the EEG in 19 normal-hearing participants while they listened to two types of stimuli differing in level of semantic context: Matrix sentences, 5-word sentences containing a proper name, verb, numeral, adjective and object with 10 options per word category presented randomly, and a semantically coherent story. Each stimulus was presented at different levels of speech understanding by adding speech weighted noise. To investigate the brain responses, we analyzed neural tracking of the speech envelope because the speech envelope is known to be essential for speech understanding and can be reconstructed from the electroencephalogram (EEG) in response to running speech. The speech envelope was reconstructed from the EEG in both the delta and the theta band with the use of a linear decoder and then correlated with the real speech envelope.

For both stimulus types and filter bands the correlation between the speech envelope and the reconstructed envelope increased with increasing speech understanding. In addition, correlations were higher for stimuli where semantic context was available. These results indicate that neural envelope tracking is more than the encoding of acoustic information and it can be enhanced by top-down processing of semantic context and speech understanding. These findings suggest that a stimulus with low semantic context should be used for our objective measure to minimize the effects of top-down processing of semantic context.
processing for both attended and unattended talkers will be evaluated from the respective temporal response functions (TRFs) and speech envelope reconstructions. With these measures, we expect to be able to track a participant’s selective attention performance over time as they grow fatigued, validating their fatigue state with our biomarkers. We will compare normal and hearing-impaired groups to assess the dynamics of how hearing loss affects selective attention during fatigue onset. Preliminary results are consistent with recent literature (e.g. Kramer S et al.) on how effortful listening modulates pupil size and alpha power. Additionally, due to its ecologically valid and dynamic nature, our paradigm also reveals the temporal interactions – and thereby potentially dissociates the underlying mechanisms – of attention, effort, and fatigue. [Work supported by Oculus Research and Starkey Hearing Technologies.]

11:50 AM  **B2-O-3: Segregation benefit from deep neural networks assessed via speech recognition and physiological measures**

*Lars Bramsløw¹, Umaer Hanif², Rikke Rossing¹, Gaurav Naithani², Tuomas Virtanen², Carina Graversen¹, Dorothea Wendt¹, Thomas Lunner¹, and Niels Henrik Pontoppidan¹,³*

¹ Eriksholm Research Centre  
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Deep neural networks (DNN) have successfully been applied for single-channel (monaural) separation of speech from noise, babble and single competing talkers (e.g. Healy et al., 2017), in listening situations where little or no spatial benefit can be expected. In everyday life, it is common that the listener wants to attend to two messages spoken simultaneously and switch attention at any time. For hearing aid applications, low latency of the signal processing is important, we have therefore developed a DNN separation algorithm with a latency of 8 ms (Naithani et al., 2017) for separating two known voices from a mixture. In a series of experiments, we have documented the benefits from this DNN algorithm for separating two competing voices from a mixture and presenting the two outputs to the two ears (dichotic). A two-talker competing voices listening test based on the Danish HINT sentences using three male and three female talkers was used with 15 hearing-impaired listeners. It showed a 13%-point benefit from DNN separation and dichotic presentation over the unprocessed mixture (p < 0.003) (Bramsløw et al., 2018). In a second experiment, the benefit was assessed using running speech and physiological outcome measures. The same spatial conditions were presented to twenty hearing-impaired listeners in an attention switching paradigm, while recording 64-channel EEG scalp potentials (Obleser and Weisz, 2012) and pupil dilation (Wendt et al., 2017). Preliminary spectral analysis of the EEG recording show that DNN separation gave same result as the ideal separation and the unprocessed (summed) signals showed a different response. The combined results from EEG spectral analysis, receptive fields and attention monitoring plus cognitive load will be presented together with results from analysis of pupil dilation caused by the challenge of competing voices listening and attention switching.
B2-P-01: Intersubject variability of phase-locked activity to a wide range of modulation frequencies assessed with auditory steady-state responses

Robin Gransier, Hofmann Michael, Marc Moonen, Astrid van Wieringen, and Jan Wouters
KU Leuven

Auditory steady-state responses (ASSRs) are phase-locked electrophysiological responses that are typically evoked by repetitive varying sounds, such as click trains and/or amplitude modulated (AM) sounds. ASSRs to AM sounds with modulations within the 0.5-100 Hz range are of potential interest for the objective assessment of hearing aids. Amplitude modulations within this range are also basic temporal components present in the envelope of speech. Temporal modulations in the speech envelope are important for speech perception, and failure of the auditory pathway to encode these modulations is associated with degraded speech perception. Assessment of hearing aid fitting with ASSRs can give potential insight in how these temporal modulations are transferred by the hearing aid and processed in the auditory pathway. For the clinical applicability of such an objective measure it is important to know the temporal modulation transfer function of the ASSR (TMFT_ASSR), and how the TMFT_ASSR differs across subjects.

We assessed the TMFT_ASSR to modulation frequencies between 0.5 and 100 Hz with a high resolution in 25 normal-hearing young adults. Seventy modulation frequencies were used to assess the full TMFT_ASSR. Each modulation frequency specific ASSR was evoked with a one-octave white-noise band centered at 1 kHz, modulated with a 100% depth, and presented at 70 dB SPL. Electroencephalography was used to record the ASSRs. The recording time per modulation frequency was 5.12 min. Our measurement paradigm allowed a detailed analysis of the intersubject variability of the TMFT_ASSR due to its long recording durations at subject level (~6 hours of effective recording time per TMFT_ASSR per subject).

Our results show a large intersubject variability in TMFT_ASSR across the 0.5-100 Hz modulation frequency range. We found that ASSRs evoked with modulation frequencies < 20 Hz, which predominately originate from cortical regions of the auditory pathway show the largest variability. Although, TMFTs_ASSR of all subjects had a prominent peak to modulation frequencies within the 35—55 Hz range, the modulation frequency that resulted in the highest ASSR amplitude differed from 40—50 Hz across subjects. ASSRs to modulation frequencies ranging from 60—100 Hz, which are used clinically to assess hearing thresholds, showed a large variation in response patterns across ipsilateral and contralateral recording electrodes. At the conference we will present the intersubject differences in the TMFT_ASSR in detail. In addition, we will discuss the effect of the intersubject differences on the utility of an ASSR based objective assessment of auditory functioning with a hearing aid.

B2-P-02: Peripheral auditory-evoked responses obtained with the Ear-EEG

Petteri Hyvärinen, Søren A. Fuglsang, Jonatan Märcher-Rörsted, and Jens Hjortkjær
Technical University of Denmark

In- and around-the-ear electroencephalography (EEG) electrodes have recently been employed in a number of applications aiming to investigate auditory-evoked activity with a setup that would be possible to integrate in a hearing aid. The feasibility of using these electrode setups configurations has mainly been assessed in relation to cortical responses, whereas the possibility of measuring peripheral function with the same equipment has not received similar
attention. However, the ability to obtain auditory brainstem responses (ABRs) or perform extra-tympanic electrocochleography (ET ECoG) with a user-operated solution could allow automated estimation of auditory function outside the clinic. This would open up new possibilities for auditory diagnostics through repeated at-home measurements.

In particular, the Ear-EEG setup where electrodes are placed inside the ear canal has the recording locations optimally positioned for the purpose of recording peripheral neural activity, originating primarily from the auditory nerve. For example, the brainstem frequency-following response (FFR) signal has been shown to be strongest when measuring with a bipolar channel between the two ear canals. Also, the traditional montage for ET ECoG uses one electrode in the ear canal, referenced to an ipsilateral electrode.

In the current study, we recorded auditory peripheral responses to 100-μs clicks and 80-ms tone bursts using three different electrode types: 1) traditional scalp electrodes, 2) TIPtrode electrodes, and 3) Ear-EEG electrodes. The scalp electrodes were placed at Fz, TP9 and TP10 positions, and the TIPtrode and Ear-EEG electrodes in the ear canals. Stimuli were presented monaurally with ER-3A insert headphones. EEG was recorded with a BioSemi ActiveTwo system at a sampling rate of 16 kHz.

This poster presents the first pilot results from the current study. The quality of recorded signals and the response morphologies are compared between the different electrode setups in order to evaluate the feasibility of the proposed Ear-EEG approach.

B2-P-03: Speech-auditory brainstem responses in individuals with hearing loss: Effects of aiding and background noise
Ghada Binkhamis1, Martin O’Driscoll2, and Karolina Kluk1
1 University of Manchester
2 Manchester University Hospitals NHS Foundation Trust

Introduction and Objectives: Objective outcome measures that can be applied in clinical audiology for young hearing aid users are limited. One of the possible measures could be the Auditory Brainstem Response to short consonant vowel speech stimuli (Speech-ABRs) designed to assess access to speech in hearing aid users. The aim of the study was to investigate the effect of aiding (with and without hearing aid) and background (quiet versus noise) on speech-ABR peak latencies and amplitudes in adults with a bilateral sensorineural hearing loss.

Methods: Speech-ABRs evoked by a 40-ms [da] were recorded from 60 adults with acquired unilateral sensorineural hearing loss (Age: MEAN = 49.87, SD = 9.43, Range = 18 – 59 years, hearing thresholds: < 70dBHL from 250Hz to 2000Hz) via loudspeaker stimulus presentation. Recordings were conducted with and without a hearing aid (aided versus unaided); in quiet and in 2-talker babble background noise using a two-channel vertical electrode montage.

Results: We found a statistically significant effect of aiding on speech-ABR peak latencies and amplitudes (p < 0.01) – with earlier latencies and larger amplitudes in the aided condition compared to the unaided condition. A statistically significant effect of background on speech-ABR peak amplitudes (p < 0.01) in both the aided and unaided conditions – with smaller amplitudes in background noise compared to quiet. And no significant effect of background on speech-ABR peak latencies in both: aided and unaided conditions.

Conclusions: Speech-ABRs evoked by 40ms [da] may potentially have clinical applications as an objective outcome measure to assess access to speech through hearing aids. [This research was funded by the Engineering and Physical Sciences Research Council, UK (EP/M026728/1), the Medical Research Council, UK (MR/L003589/1), Saudi Arabian Ministry of Education and supported by the NIHR Manchester Biomedical Research Centre.]

B2-P-04: Effects of noise exposure on young adults with normal audiometric hearing
Karolina Kluk1, Garreth Prendergast1, Hannah Guest1, Rebecca Millman1, Kevin Munro1, Rebecca Dewey2, Susan Francis3, Deborah Hall2, Chris Plack1
1 The University of Manchester
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Results from rodent models suggest that noise exposure can cause loss of synapses between inner hair cells and auditory nerve fibers (cochlear synaptopathy) without affecting threshold sensitivity. However, in a series of studies involving adults aged 18 to 35 with normal audiograms and a wide range of lifetime noise exposures we have found no evidence for an effect of noise exposure on auditory brainstem response amplitude (including wave I), or envelope following response amplitude. We have also found no evidence for an effect of noise exposure on behavioral
measures including frequency discrimination, intensity discrimination, interaural phase discrimination, modulation detection, sound localisation, musical harmony perception, and speech-in-noise identification. In a companion fMRI study, we have also found no evidence for noise-induced changes to the central auditory pathways. It seems likely that humans are less susceptible to noise-induced synaptopathy than rodents. Noise-induced synaptopathy may only be a significant cause of hearing deficits in humans with extreme exposures, and may always occur in combination with a high-frequency audiometric loss. [This research was funded by the Medical Research Council, UK (MR/L003589/1), an Action on Hearing Loss studentship - the Marston Family Foundation, and supported by the NIHR Manchester Biomedical Research Centre.]

**B2-P-05: Is it enough to use the objective hearing test for pediatric hearing rehabilitation? The importance of behavioral pure tone audiometry through auditory training with hearing aids**

_Hye Jin Lim, Ji Hye Kim, You- Ree Shin, Seoung-Cheon Bae, Kyoung Ray Moon, and Young-Myoun Chun_  
Soree Ear Clinic

The diagnosis of hearing loss in infants and young children is generally based on the objective methods (auditory brainstem response (ABR) or auditory steady state response (ASSR)), and auditory rehabilitation such as hearing aids or cochlear implant were followed promptly according to the degree of hearing loss. However, hearing aids fitting is limited due to lack of the exact hearing threshold throughout the frequencies from objective hearing test. Thus, we have tried to get behavioral pure tone audiometry (B-PTA) results combined with auditory training wearing hearing aids. The purpose of this study was to compare the results between B-PTA and objective measurements with clinical significance. Methods: One hundred twenty-two patients under aged 3-year-old who underwent objective tests and reliable B-PTA from January 2013 to March 2017 in Soree Ear Clinic were enrolled to this study. For comparison ABR to B-PTA, the low tone hearing thresholds in B-PTA were evaluated in 27 patients (54 ears) with bilateral no response (NR) on ABR. And, for comparison of ASSR to B-PTA, the correlation between two tests was analyzed in each frequency in 65 patients (130 ears). Results: Regarding the mean threshold of 125Hz, 250Hz and 500Hz of B-PTA in ABR NR patients, 35 ears (65%) were less than 90dBHL (<50dB:3(5.6%), 50~60dB:6(11.1%), 70~80dB:9(16.7%), 80~90dB:10(18.6%) ears each). Sixteen (29.6%) ears were greater than 90dB, and only 3 (5.6%) ears were NR in low frequencies at B-PTA. The correlation between ASSR and B-PTA in each frequency was significant (acceptable or good correlation: Cronbach’s alpha coefficient: 250Hz=0.767, 500Hz=0.790,KHz=0.833, 2KHz=0.861, 4KHz=0.852 and 250-4KHz average= 0.873). In low frequencies, the coefficient is slightly lower than mid-high frequencies. In addition, the threshold difference between ASSR and B-PTA, over than 20dB, were 70% in 250Hz, 55% in 500Hz, 33% in 1kHz, 26% in 2kHz, 25% in 4kHz, and 36% in average. Conclusion: Residual functional hearing existed with various range even in cases with NR on ABR. Even though ASSR thresholds were significantly correlated with B-PTA in all frequencies, many cases had differences more than 20dB between the results of ASSR and B-PTA. Therefore, in addition to ABR and/or ASSR, obtaining the B-PTA through auditory training with hearing aids is important for the best fitting of hearing aids and choosing strategies for the combined use of cochlear implants and hearing aids for the better auditory rehabilitation outcome.

**B2-P-06: Listening effort and alpha power: Effect of directionality in hearing aid users**

_Pragati Rao Mandikal Vasuki, Jason Galster, and Jeff Crukley_  
Starkey Hearing Technologies

Listening in the presence of background noise is the most challenging listening condition faced by individuals with hearing impairment (HI). When compared to listeners with normal hearing, those with HI expend more listening effort when listening in noise; this effort results in fatigue and loss of mental focus towards the end of a day. Moreover, listeners with HI demonstrate a large degree of individual variability on listening in noise tasks, thus also suggesting also large degrees of individual variability in listening effort. Measurement of listening effort, while controlling for individual variability in performance levels presents a formidable challenge to researchers. Dual-task paradigms have frequently been used to measure listening effort. In these paradigms, responses on a secondary task, performed concurrently with a primary listening task, are slower and more error-prone as listening effort increases. One criticism of this approach is that task performance maybe be dependent on, and thus confounded by attention. Recently, the use of physiological signals such as electroencephalography (EEG) to measure listening effort have
shown promise in assessing listening efforts without the confounds dual-task paradigms. Hearing aid users experience reduced listening effort when using signal processing strategies such as directional microphones (Desjardins, 2016; Picou et al., 2017). However, the effects of directional microphone use on EEG measures of listening effort while controlling for individual performance levels has yet to be reported. Purpose: This study investigated the effect of directionality and individual performance levels on listening effort measured with EEG during a sentence recognition task in hearing aid users. Design: Signal-to-noise ratios (SNRs) for sentence recognition at two performance levels (50% & 80%) were individually measured using an adaptive task (Keidser et al., 2013). EEG signals were recorded during sentence recognition at these measured SNRs under two microphone conditions (omni- and fixed-directional). Participants also reported subjective listening effort ratings for each performance level and microphone condition using a 7-point scale. Additionally, participants’ working memory capacity (WMC) was assessed to examine the relationship between WMC and listening effort. Results & Discussion: Single-trial alpha power was extracted from the EEG signal during sentence presentation and during a baseline interval prior to the sentence onset. The change in alpha power during sentence presentation was used to index listening effort. This poster will present and discuss the relationships between alpha power, subjective listening effort, and WMC Implications for real-time assessment of listening effort with EEG will also be discussed.

B2-P-07: Individual differences in the strength of middle-ear-muscle reflex in young adults with normal hearing and varying degree of noise exposure

Chhayakant Patro, Heather A. Kreft, and Magdalena Wojtczak
University of Minnesota

A relatively large number of individuals seeking audiological assessment for deficits in speech understanding under adverse listening conditions have clinically normal audiograms. One reason for such deficits in the absence of audiometric hearing loss may be permanent loss of synapses between cochlear inner hair cells and auditory nerve fibers. The “cochlear synaptopathy” has been demonstrated in animal models using histopathological assessment. Animal studies also reported strong relationships between the synaptic loss and noninvasive physiological measures that can be used in humans. Unfortunately, electrophysiological measures of auditory brainstem responses in humans are strongly affected by non-auditory factors which contribute to a large inter-individual variability that may obscure any differences due to synaptopathy. One of the noninvasive physiological measures, the middle-ear-muscle reflex (MEMR) strength, has emerged as a promising marker of cochlear synaptopathy. The reflex has been shown to be significantly reduced in noise-exposed animals with cochlear synaptopathy and in humans with noise-induced tinnitus and normal audiometric thresholds. Despite a large significant effect of tinnitus on MEMR strength, the MEMR in over 20% of individuals in the control group was found to be as weak as that for the tinnitus group. The aim of this study was to investigate whether individual differences in MEMR strength in young normal-hearing adults with different self-reported noise exposure correlate with electrophysiological and perceptual measures of temporal processing thought to be affected by cochlear synaptopathy. Twenty subjects (mean age - 22.8 years, standard deviation – 4.8 years; thirteen females, seven males) participated in the study. All subjects had normal hearing of 15 dB hearing level (HL) or better, at octave frequencies between 250 Hz and 8000 Hz. A wideband measure of the MEMR was used to probe individual differences in MEMR strength and to investigate the relationship between the reflex and measures that have been used in previous studies to investigate the presence of cochlear synaptopathy in humans. These measures included envelope-following responses to high-frequency tones modulated in amplitude at high rates, and behavioral measures of amplitude-modulation detection and the detection of an interaural envelope phase difference. The behavioral measures were obtained in quiet and in noise. Large inter-individual differences were found in all measures but no correlation was found to be significant. The lack of significant correlations will be discussed in terms of orthogonal factors affecting the different measures used in the study.

B2-P-08: Application of real time recurrent neural networks for estimating probe tube insertion depth

Jonathan Pietrobon¹, John Pumphord¹, Paula Folkeard², and Chris McInerney¹
¹ Audioscan
² University of Western Ontario
Real ear measurement is widely considered to be best practice for hearing instrument verification by a variety of professional organizations around the world. Real ear measurements are performed by inserting a probe tube attached to a probe microphone into the ear canal of a subject and measuring the acoustic response of a sound source external to the ear from an opening at the tip of the probe tube. The accuracy of such measures is dependent on the position of the probe tube relative to the tympanic membrane. Accurately positioning the probe tube is an important consideration when performing real ear measurements. As well, probe tube insertion can be a challenging and time-consuming task which some clinicians cite as a reason for not performing real ear measurements. Various methods are currently used to estimate probe tube insertion depth including visually assisted positioning which considers the anatomy of the external ear along with the use of an otoscope or acoustically assisted positioning. Current acoustic methods rely on standing wave theory using either visual interpretation of a spectrum or a series of repeated measurements where the probe tube is moved by some amount in-between each measurement. This submitted work seeks to develop an accurate real-time estimate of probe tube insertion depth that works seamlessly with clinical practice to improve the accuracy of and reduce barriers to performing real ear measurements.

Recordings of probe tube insertions were made with 100 adult ears using an AudioScan Verifit 2. Subjects were presented with a shaped noise signal from a loudspeaker and measurements from both the on-cheek reference microphone and the in-ear probe tube microphone were made while a clinician performed a probe tube insertion. These recordings were analyzed using non-causal signal processing techniques to estimate the depth of the probe tube insertion at 128 millisecond intervals. A recurrent neural network model was trained to predict the labelled probe tube insertion depth using the short-time real ear unaided gain in fractional octave bands as inputs. K-fold cross validation of the model suggests that it is successful at robustly predicting probe tube depth for unseen recordings. When integrated into a clinical tool, the model provides good test-retest accuracy of real ear measurements and reduces the time required to complete probe tube insertion. Real ear measurements with the probe tube positioned visually by an experienced clinician and using the clinical tool are compared.

B2-P-09: Ultra hearing: Selectivity and safety of amplitude-modulated ultrasound stimulation for a new hearing technology

Gerardo Rodriguez, John Basile, Hongsun Guo, Cory Gloeckner, and Hubert Lim
University of Minnesota – Twin Cities

Background: Recently, our lab has shown that ultrasound applied to the head activates the auditory system likely via vibration of cerebrospinal and cochlear fluids (Guo et al., Neuron, 2018, In Press). This discovery of noninvasive transcranial ultrasound activation of the peripheral auditory pathway opens up the possibility for a new type of hearing technology, which can bypass damaged outer and middle ear regions, does not require a device inserted into or occluding the ear canal, and can avoid acoustic feedback issues. We are investigating the neural selectivity of ultrasound activation of the auditory system via amplitude modulated ultrasound pulses. We are also investigating the safety of different ultrasound parameters that are relevant for hearing applications.

Materials and Methods: We positioned 32-site electrode arrays in the central nucleus of the inferior colliculus (ICC) of ketamine-anesthetized guinea pigs. Ultrasound (220 kHz) pulses modulated by low-frequency tones (1-40 kHz) and varying pressure levels (10-100 kPa) were presented on various regions of the skull and the exposed brain (coupled with degassed agarose). The transducer was also decoupled from the animal to confirm there were no air-conducted sound effects. To investigate the stimulation safety, we presented ultrasound (100 or 220 kHz) to isoflurane-anesthetized guinea pigs for 5 hours each day over a period of 5 days with different pressures (100-800kPa), duty cycles (5-80%), and sonication durations (.05-9.6 s). We then performed histological analysis to characterize any brain tissue damage. Results: Initial results demonstrate that the auditory midbrain can be selectively activated in a tonotopic pattern with different modulation frequencies, and higher stimulation intensities elicit broader activation across ICC. Histological analysis also suggests that ultrasound pressures ranging from 100 to 800 kPa do not induce noticeable damage of the brain at low duty cycles and durations. However, further work will need to be performed to characterize the safety limits of ultrasound pressure, duty cycle, and duration interactions for more complex stimulation patterns.

Conclusions: Ultrasound applied to the head appears to achieve safe and frequency-specific activation of the auditory system at very low ultrasound intensity levels. Future studies will evaluate if complex sound
Spatial auditory attention (SAA) is an aspect of the auditory systems ability to solve the cocktail party problem. There is a cortical representation of a listeners attended soundstream and stimulus reconstruction can be used as a tool to decode SAA. The aim of this study was to evaluate stimulus reconstruction from single electroencephalographic (EEG) trials in a realistic acoustic environment and to investigate its limitations. Within a free field environment consisting out of four loudspeakers, ten participants had to alternately focus their SAA on one out of four loudspeakers. Those were equidistantly arranged in a half circle (-90°, -30°, +30° and +90°). To ensure that the participants were able to solve the listening task, they had to answer content related multiple choice questions at the end of each of the 24 experimental trials. On the basis of the acquired 128 channel EEG data, decoders were computed to reconstruct speech envelopes. The approach’s basic idea is that the cortex acts like a linear time invariant system mapping input to a certain output. The ongoing EEG activity resulting from ongoing stimulation can be interpreted as a linear convolution. The instantaneous neural activity is the result of a convolution of the acoustic stimulation with an unknown, channel-specific temporal response function (TRF). The TRF is a filter describing the transformation of the stimulus to the EEG activity. Here, it is more interesting that the described can be used the other way around, i.e., to reconstruct the stimulus from the EEG. For each participant and trial, the speech envelope was reconstructed and Pearsans correlation was used to compare the reconstructed speech envelope with the four actual speech envelopes. The speech envelope showing the highest correlation was determined as the attended one. If the predicted attended speech envelope matched the actual attended speech envelope, the spatial auditory attention decoding was counted as correct. The amount of correct answered multiple choice questions is 76.8 +/- 13.1% in average. It was possible to correctly decode the attended loudspeaker with up to 84% accuracy. Guessing should result in 25% accuracy (four loudspeakers), the achieved accuracy is remarkably high. The results suggest that it is possible to use the approach to decode the orientation of SAA in this multi-talker environment.

**B2-P-11: A Granger causality-based method to decode the orientation of spatial auditory attention exclusively from the EEG**

Patrick Johannes Schäfer\(^1\), Zeinab Mortezapouraghdam\(^1\), Farah I. Corona-Strauss\(^2\), Ronny Hannemann\(^3\) and Daniel J. Strauss\(^1\)

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The term cocktail party problem describes a listener’s ability to focus one out of several competing talkers. The ability to spatially direct auditory attention plays a major role in solving this problem. Current research investigates effects of auditory attention on ongoing electroencephalographic (EEG) activity caused by ongoing stimulation, i.e., real speech. It is possible to reconstruct characteristics of auditory stimulation from the EEG and it has been shown that stimulus reconstruction is sensitive to selective auditory attention and can be used to decode auditory attention. The disadvantage of this approach is that it is not possible to decode auditory attention directly from EEG – EEG and physical characteristics of the acoustic environment are necessary. Here, a new way to decode the orientation of spatial auditory attention (SAA) based exclusively on ongoing EEG data is presented. The method is based on Granger causality (GC), best known for expressing the functional directivity between different processes. Using GC, it can be determined if the past information of a second process Y contains useful information to predict the future of process X higher than of using only the past information of X. In other words, the inclusion of past information of Y and X yield a better prediction of X, than not using Y. 17 subjects took part in this study and had to alternately focus one out of two competing loud speakers (-90° and +90°). The acoustic stimulation, i.e., several different podcasts, was played via those loudspeakers. The acquired and pre-processed EEG data was used to compute GC coefficients between different electrodes of relevance (selected in
the way that they cover areas of interest - primary auditory cortex, Wernicke's area and the "dorsal stream"). The resulting GC coefficients show patterns on how dependencies between electrodes are orientated during the two listening conditions. Those patterns were used to classify between the different directions of SAA. As classifier a support vector machine (SVM) algorithm with a linear kernel function was used. The proposed classification method was used individually as well as over the total set of participants. In both cases the results show that the proposed approach is able to reliably detect the orientation of SAA, i.e., 84 +/- 2.6% and 70.45 +/- 0.74% respectively. It is concluded that it is possible to decode SAA exclusively from ongoing EEG activity and that there could exist a general pattern within the recorded EEG caused by SAA.

**B2-P-12: Impact of hearing aid noise reduction algorithms on speech-evoked auditory brainstem response**

*Hye Yoon Seol, Stayeon Park, Soojin Kang, Young Sang Cho, Sung Hwa Hong and Il Joon Moon
Samsung Medical Center*

When diagnosed with hearing loss, hearing aids (HAs) are typically recommended as an option. HAs amplify sounds to increase speech intelligibility and they contain other features, such as noise reduction (NR) and frequency lowering, to help HA users hear better in noisy situations. As speech comprehension entails both peripheral and central processing, it is crucial to understand how the brain integrates the acoustic inputs. The auditory brainstem response to complex sounds (cABR) has been gaining more traction as it is an objective and noninvasive way to assess neural encoding with real-world stimuli rather than clicks and tone bursts. The potential role of cABR in improving HA outcomes has been explored in literature. The purpose of this study is to investigate the neural representation of a /da/ stimulus in the auditory system of individuals with normal hearing (NH) and those with HAs and the impact of NR on cABR responses. Twenty individuals with NH and 29 HA users participated in this study and completed pure-tone audiometry, the Korean version of Hearing in Noise Test (K-HINT), and cABR. The NH group was tested in /da/ only and /da/ with white noise (WN) conditions while the HA group was tested in four conditions: /da/ only, /da/ WN, /da/ WN NR ON, and /da/ WN NR OFF. Both groups were tested in two signal-to-ratio (SNR) conditions: 0 and +5dB. Our results revealed significant differences between both groups for first harmonics and fundamental frequency in /da/ only and /da/ WN conditions. Although the presence of noise reduced cABR amplitudes in both groups, NR and SNR did not have any impact on the cABR responses. K-HINT results showed that the reception thresholds for sentences were similar regardless of the activation of the feature in both quiet and front noise conditions for HA users. Statistical analysis also revealed no significant difference between the aided NR ON and aided NR OFF conditions for HA users. Findings of this study are consistent with previous literature that with reduced cABR amplitude, sounds are less represented in individuals’ auditory system. Since cABR results are consistent with behavioral test results, they could become a powerful counseling tool for HA users. As a result, HA users will have more realistic expectations, such as acknowledging that speech understanding in noise will still be challenging even with HAs and are encouraged to actively use communication strategies.

**B2-P-13: Listening effort and speech intelligibility measurement for multiple subjects simultaneously in complex listening scenarios**

*Larissa Taylor\(^1\), Henry Luo\(^2\), and Ian Bruce\(^1\)

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The experiences of hearing aid users in their daily lives do not show the same amount of benefit from the noise reduction or directional algorithms that has been found in lab-based studies looking at listening effort and speech intelligibility. This is because most of the testing has been done in scenarios designed to maximize the benefit of the directionality and not conditions found in the more complex environments hearing aid users regularly encounter. This study used the McMaster LIVElab to test the effects of fixed versus adaptive directionality processing on speech intelligibility and listening effort simultaneously in a group of subjects situated in a controlled but complex listening environment.

Two experiments were performed to measure listening effort and speech intelligibility in background noise and reverberation. The first measured listening effort during a live play performance, while the second measured speech intelligibility and listening effort when listening to recorded sentences played back from different directions. Both experiments took place in the presence of background noise played from all directions in the LIVElab to make listening effortful for the normal hearing subjects. Subjective
ratings of listening effort were collected as well as ECG and EEG data for objective physiological measures of listening effort. The subjective results showed that young normal hearing subjects have an increased speech intelligibility and reduced listening effort with the Unitron SpeechPro and Spatial Awareness compared to fixed beamformer directionality in a number of the listening scenarios investigated. There are several methods described in literature for calculating listening effort from ECG and EEG data. Since the subjective and physiological data were collected simultaneously, our goal is to determine if one of these objective measures gives listening effort results similar to the subjective ratings. [This research was funded by Unitron, a Sonova brand. We thank the LIVELAB and Unitron staff who helped make these experiments possible and the subjects who participated in the experiments.]

**B2-P-14: Effect of envelope shape on the amplitude of the following response**

*Jana Van Canneyt, Michael Hofmann, Jan Wouters and Tom Francart*
*KU Leuven*

**Background:** Envelope following responses (EFRs) are auditory evoked potentials that reflect dynamic phase-locked activity to temporal features of the evoking stimulus. EFRs are valuable in research on supra-threshold hearing but also clinically, to objectively determine hearing thresholds. EFRs can be measured with a range of stimuli, but most common is the sinusoidally amplitude modulated tone.

**Objective:** To use EFRs for hearing diagnostics or hearing aid fitting, short measurement time and resilience to high measurement noise is crucial. This can be achieved by optimizing stimulus parameters to obtain high response amplitudes. Hence, we investigated the effect of stimulus envelope shape on EFR amplitude.

**Methods:** We used trapezoidal modulated tones (modulation frequency = 40 or 80 Hz) for which the envelope shape is defined by attack time, hold time, decay time and off time. The effect of these four parameters on the EFR was investigated with EFR measurements (EEG, 64 electrodes), and with a model of the auditory periphery [Bruce et al., Hearing Research, 360, 40-54 (2018)].

**Results:** Results from the neural model showed that response modulation depth is mostly influenced by off time and decay time. There was a linear relation between longer off time and lower minimum firing rate (i.e. higher response modulation depth). However, short decay times (< 7.5 ms) produced large neural activity that 'filled up' the off time, causing exponentially higher minimum firing rates (i.e. lower response modulation depth). Contrary to expectation, attack time did not influence maximum firing rate. First results from the EFR measurements confirm the effect of off time and decay time on EFR amplitude.

**Conclusions:** Decay time and off time have significant influence on EFR amplitude. The largest EFR amplitudes can be obtained for a decay time > 7.5 ms combined with a sufficiently long off time. [This research is funded by FWO (Research foundation Flanders) within the framework of the TBM-project LUISTER (T002216N) and jointly by Cochlear Ltd. and Flanders Innovation & Entrepreneurship (formerly IWT), project 50432. Financial support was also provided by an SB PhD fellowship from FWO to Jana Van Canneyt.]

**B2-P-15: Impact of SNR, hearing impairment, and hearing aid technology on listening effort as indicated by the pupillary response**

*Dorothea Wendl¹, Adriana Zekveld², Barbara Ohlenforst², Graham Naylor³, Sophia Kramer³, and Thomas Lunner³*
*¹ Eriksholm Research Centre*
*² VU University Medical Center and Amsterdam Public Health Research Institute*
*³ University of Nottingham*

Speech comprehension in everyday communication can be effortful even when speech is fully intelligible. Acoustical distortions of the incoming speech signal due to interfering background noise can make speech comprehension more effortful, especially for people with hearing impairment. Signal processing in modern hearing aids and noise reduction (NR) schemes aim to counteract the detrimental effect of noise and reduce the effort required for speech comprehension. Two different pupillometry studies will be presented that explored the impact of task demands (imposed by background noise), hearing status, and hearing-aid signal processing on listening effort in adults. The first study investigated the impact of interfering babble noise on listening effort across a broad range of SNRs within a group of listeners with normal hearing. The results suggested that the maximum pupil dilation changed across SNRs in a non-linear way. Pupil dilations were largest, interpreted as listeners allocated most cognitive resources and, thus, listening effort at performance accuracies around 50% correct speech recognition. However, with further increasing
task demands (i.e. decreasing SNRs) pupillary response decreased indicating that listeners allocated fewer resources for speech recognition. The reduced pupil dilations were interpreted as signs of ‘giving up’ due to a drop in motivation in those situations where speech recognition was most difficult. The second study examined the impact of hearing impairment and hearing-aid signal processing on listening effort across a broad range of SNRs. A clear effect of both hearing impairment and a NR scheme on the pupil dilation was observed. In listening situations with high speech intelligibility, the release of effort was smaller for listeners with hearing impairment than for listeners with normal hearing as indicated by the pupillary response. In addition, hearing-aid signal processing (i.e. NR) decreased the listening effort required for speech recognition in noise. Taken these, both studies give an insight into listening effort across a broad range of acoustic scenarios including ecologically valid listening situations.

B2-P-16: The difference between objective and subjective speech intelligibility
William M Whitmer and David McShefferty
University of Nottingham

When discussing speech intelligibility benefits, it is common to refer to the signal-to-noise ratio (SNR) where a listener’s ability to repeat the signal correctly 50% of the time (SNR50). If performance has been measured robustly, there should be objective equivalence in difficulty across any signal and noise pairs presented at a listener’s SNR50. It is reasonable to assume that the listener’s perception of difficulty will also be equivalent across stimuli presented at their respective SNR50s. We found this assumption of subjective equivalence to be false.

Twenty adult (median age of 67 years) listeners (nine female) of varying hearing ability (median better-ear average 29 dB HL) participated. In different blocks of trials, listeners first were tasked with repeating back IEEE sentences in same-spectrum or two-talker babble noise at various SNRs. Individual SNR50s were then estimated from the psychometric functions. Thereafter, listeners heard on a given trial two intervals: a sentence presented in babble and the same sentence presented in same-spectrum noise. One interval would be presented at its SNR50 and the other at its SNR50 plus an SNR increment varying from 0-8 dB in 2 dB increments. Listeners were asked to choose which sentence was clearer. All stimulus combinations and orders were counter-balanced and repeated 12 times.

The result of note was when there was a 0 dB increment (i.e., when both stimuli were presented at their SNR50). It was initially expected that listeners would choose each stimulus 50% of the time. Listeners on average, however, chose sentences in babble to be clearer 64% vs. 36% for sentences in same-spectrum noise [t(19) = 7.81; p < 0.0001]. In the rest of the conditions, this preference or “clarity gap” persisted. There was no correlation between the clarity gap and individual SNR50s nor individual differences in SNR50s. The Hearing Aid Sound Quality Index modelled the clarity gap relatively well. These results indicate a difference between objective, perceptual benefits and subjective, perceived benefits. If equivalent performance is not perceived as being equivalent in clarity across stimuli, perhaps an altogether different measure, such as effort, could yield subjective equivalence. [Work supported by the Medical Research Council (grant number U135097131) and the Chief Scientist Office of the Scottish Government.]

B2-P-17: Study of abnormal phone perception in children with reading disability
Jont Allen and Yashuo Wu
University of Illinois

Reading disability (RD) is widely viewed as a key obstacle in the development of literacy. Studies show that between 15-20% of grade-school students have RDs, and as a result many drop out of school in their early age (i.e., by high-school). According to national statistics, fifty percent of the inmates in jail cannot read. One might reasonably conclude that RD can be a ticket to jail for a significant percentage of RD children. It follows that understand the source of such RD could impact the success rate of treatment.

We shall show that the source of RD in young children (8-12 yrs) is related to inadequate phonetic non-categorical processing skills, rooted in pre-school language development. This conclusion is based on two experiments on children with documented reading disabilities: 1) a Syllable Confusion Oddball task (SCO) and 2) a Nonsense Syllable Confusion Matrix task (NSCM). In this report, we focus on the SCO task.

The SCO task tested normal-hearing RD children, having normal language function, in their ability to identify different syllable (CV, VC) from a string of
three such syllables, spoken by three different talkers, from a data base of 20 adult talkers. The consonants and vowels had 4.2 bits of entropy (19 vowels and 19 consonants), randomly selected from our data base of spoken sounds (LDC AI corpus). The average number of trials was approximately 10 thousand per child, taken over 10, 1-hour sessions.

The experimental results showed that the 10 RD children had 5 times the error compared to the reading control (RC) group. The results were analyzed using the logit procedure (R+lme4) (Jaeger, 2008).

Conclusions:

1) RD children have a significant speech perception problem in identifying nonsense syllables, despite normal pure-tone hearing and language processing ability.

2) Based on a simple model of the results, the source of this disorder appears to be a poor memory for a subset of phones, which is different for each RD child.

3) Our conclusions are at odds with previous publications which found no sign of phone impairment. However, these previous studies were limited in the number of trials and the number of phones that were tested, as well as the type and extent of trials.

Thursday, August 16

SESSION B3. Individualized Diagnostics, Fitting, and Rehabilitation: Part I

Session Chair: Stefan Launer

5:00 PM B3-O-1: Hearing aid fitting and the individual: integrating measures of dynamic range and audibility with individualized diagnostics

Susan Scollie¹, Maaike Van Eeckhoutte², Danielle Glista², Paula Folkeard², Ewan Macpherson¹, and Prudence Allen¹

¹ University of Western Ontario
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Hearing aid fitting has historically aimed to overcome the inaudibility caused by threshold elevation, using the application of gain per frequency, per ear. In modern practices, our industry has improved upon this basic goal by providing level-dependent multichannel processing with enhancements from digital signal processing to clean the signal, link ears, and steer beams. Nonetheless, we are often left with between-listener performance variation in which some listeners outperform others even under conditions of similar age, hearing status, and hearing aid type. In this presentation, I will focus on current studies underway at our centre to better understand several of the factors that may relate to variance in hearing aid outcome, including how we define the dynamic range of hearing (especially thresholds) for the purposes of hearing aid fitting, the role of auditory mapping, and current studies of supra-threshold perception of aided sound. One current focus area is the implementation of wideband amplification, including the methods used to assess, fit and verify, as well as understanding the candidacy for, and possible benefits of, wideband amplification. Results from these projects, as well as studies of loudness, preference, and verification of extended band fittings will be used to explore several factors that may relate to individual performance.
B3-O-2: Hearing aid processing strategies for listeners with different auditory profiles: Insights from the BEAR project

Mengfan Wu¹, Mouhamad El-Haj-Ali¹, Raul Sanchez-Lopez², Michal Fereczkowski², Federica Bianchi², Torsten Dau², Sébastien Santurette², and Tobias Neher¹
¹ University of Southern Denmark
² Technical University of Denmark

Background: The Better hEAring Rehabilitation (BEAR) project pursues the development and evaluation of new clinically feasible strategies for individual hearing loss diagnosis and hearing aid fitting. Two essential elements of this research are the design of a new diagnostic test battery for identifying different auditory profiles and linking those profiles to hearing aid processing strategies. The current study focused on establishing links between four auditory profiles and benefit from six hearing aid processing strategies.

Methods: Participants were 30 older individuals with bilateral mild-to-severe sensorineural hearing losses who were selected from a clinical population of hearing aid users. Speech-in-noise stimuli were generated with the help of a hearing aid simulator that included directional processing, noise reduction and dynamic range compression. Stimulus presentation was via headphones. Six hearing aid settings that differed in terms of signal-to-noise ratio (SNR) improvement and temporal and spectral speech distortions were selected for testing based on a comprehensive technical evaluation of different parameterisations of the hearing aid simulator. Speech-in-noise perception was assessed at fixed input SNRs that were selected based on individual speech reception threshold (SRT50) measurements. Participants were required to recognize five-word, low-context sentences embedded in two realistic noise backgrounds. In addition, overall preference and noise annoyance were assessed using a multiple stimulus comparison paradigm.

Results: We hypothesize that the perceptual outcomes from the six hearing aid settings will differ across listeners with different auditory profiles. More specifically, we expect listeners showing high sensitivity to temporal and spectral differences to perform best with and/or to favor hearing aid settings that preserve those cues. In contrast, we expect listeners showing low sensitivity to temporal and spectral differences to perform best with and/or to favor settings that maximize SNR improvement, independent of any additional speech distortions. Altogether, we anticipate that these findings will provide the basis for more individualized fitting strategies to be implemented in wearable hearing aids.

B3-O-3: Novel hearing instrument and fitting protocol for treating hyperacusis

David Eddins¹, Steve Armstrong², and Craig Formby³
¹ University of South Florida
² SoundsGood Labs
³ University of Alabama

Patients who suffer a debilitating intolerance to the loudness of everyday sounds, a condition known as hyperacusis, present a unique treatment challenge. Such patients often present in the clinic wearing earplugs (EPs) to limit offending sound exposures. We and others have shown chronic use of EPs increases auditory gain and exacerbates the hyperacusic condition, rendering the EP wearer even more sensitive to loud sounds. In contrast, we have shown chronic use of ear-level sound generators (SGs), which produce low-level noise, reduce auditory gain, thus, increasing tolerance for loud sounds. This presentation will describe a novel transitional device incorporating both EPs and SGs, and the associated fitting protocol, for treating the severely hyperacusic patient. To meet the patient’s pre-treatment needs, a deeply seated and acoustically sealed in-the-ear mold offers maximum sound attenuation. This mold includes a heat-activated stint that expands at body temperature to augment the normal seal, functioning as a high-quality, custom EP. A miniature behind-the-ear hearing device is connected to the earmold via slim tube. The device has four key functions
in addition to the attenuation provided by the EP. An on-board SG creates a low-level, spectrally-shaped noise, is configurable for individual patients, and reduces auditory gain and increase sound tolerance. As the SG induces loudness tolerance change, amplification approaches unity gain over time to overcome the maladaptive plasticity associated with earplugging. Simultaneously, output limiting (loudness suppression) reduces the exposure to loud, offending sounds. If the patient has aidable hearing loss, then the device can function as a fully-featured hearing aid.

A fitting protocol has been developed so that the patient can realize the desired benefit of this novel treatment device. During the first fit, real-ear measures quantify unaided gain, occluded gain, the noise response, and the aided unity gain needed to overcome earplugging. Output limiting, imposed under conditions of unity gain, minimizes exposure to loud sounds while providing access to soft and comfortably loud sounds typically attenuated by an EP that otherwise exacerbates hyperacusis. A real-ear noise response is measured and adjusted to the desired spectral shape, activated. The patient undergoes counselling on use, care, goals, and expectations that the low-level noise will enhance sound tolerance. On subsequent visits, the resulting SG-induced increases in loudness tolerance determine the release of loudness suppression and the transition of the patient from EPs to normal, device-free audition, ultimately offering an effective treatment for debilitating hyperacusis.

[Work Supported by NIDCD R21 DC015054.]

POSTER SESSION I

**Paired to Sessions B1, B2, and B3**

**Thursday 9:40 AM – 11:10 AM, 8:40 PM – 10:00 PM**

Posters for Session I should be put up by 8:00 AM Thursday, August 16, and taken down after 10:00 PM Thursday, August 16. Presenters should be at their posters from 9:40 AM – 11:10 AM

**B3-P-01: Explaining fine-tuning gain differences for subjects with similar audiograms**

Dirk Oetting1, Michael Schulte2, Wouter Dreschler3, Monique Boymans4, Birger Kollmeier5, Stephan Ewert5, and Jörg-Hendrik Bach1

1 HörTech gGmbH & Hearing4all, Oldenburg
2 Hörzentrum Oldenburg
3 Academic Medical Center, Amsterdam
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5 Hearing4All & Universität Oldenburg

Hearing aid fittings typically use prescriptive formulas based on the audiogram. In everyday use, binaural, suprathreshold sounds play a significantly more important role than monaural near-threshold sounds. Keidser et al (2012) found that the spread of individual gains after fine-tuning is large and cannot be explained by the audiogram. One reason may lie in an effect known as binaural, broadband loudness summation. A large spread of individual loudness perception when measuring binaural broadband signals was found in subjects with similar hearing thresholds (trueLOUDNESS, Oetting et al. 2016). It is an open question if and how much this effect may explain individual spread of gain curves, and whether there is a relationship between loudness-based gain predictions and user preferences (e.g. power-users vs loudness sensitive subjects). In the present study, we compared individual gains after fine-tuning for input signals at 65 dB SPL. Gains were assessed by calculating the required gains for binaural broadband loudness normalization at 65 dB SPL according to the procedure proposed by Oetting et al (2017). Gains for monaural loudness normalization are corrected to achieve normal-hearing binaural broadband loudness perception. Furthermore, candidates whose trueLOUDNESS-based gains are remarkably higher than the threshold-based expectations should be able to be classified as so-called power-users, and those with remarkably lower trueLOUDNESS gains as loudness-sensitive subjects. To this end, we provided hearing aid acousticians with a questionnaire to assess their rating of the patients in terms of loudness perception. The
acousticians did not have access to the predicted gains of the patients at that time. The individual spread of gains based on trueLOUDNESS is comparable to fine-tuning values reported in the literature (Keidser et al, 2012). We also found a relation between real-world experiences of hearing aid users (loudness sensitive/power-junkies) to the predicted trueLOUDNESS gains. The results showed significant potential to improve hearing-aid first fit procedures by using suprathreshold measurements. The suprathreshold measurements should contain binaural and broadband test-signals. Using loudness-scaling measurements, prescription-based first fits may get significant improvements, at least for user groups at the extremes (i.e., power junkies and loudness sensitive users). The additional binaural broadband measurements have the potential to reduce fine-tuning steps and make target gain functions more reliable.

**B3-P-02: Application of open-source speech processing platforms to independent hearing-aid research: Two examples**

Arthur Boothroyd\(^1\), Gregory Hobbs\(^1\), Carol Mackersie\(^1\), and Harinath Garudadri\(^2\)

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An NIDCD initiative is funding development of open-source speech processing platforms to facilitate independent hearing-aid research. The long term goal is to improve accessibility and quality of hearing health-care. This presentation offers two examples of ongoing hearing-aid-related research at San Diego State University (SDSU) involving the platform under development at the University of California in San Diego (UCSD). One example is a study of the efficacy of, and candidacy for, hearing-aid user self-adjustment. The other is evaluation of a self-administered test for use during self-adjustment. The UCSD platform uses time-domain filtering to create six sub-bands. In each band, gain, amplitude compression, and output limiting are controlled independently. Control is accomplished wirelessly from an Android tablet. The option for audio input and output via hearing-aid transducers is included, along with the necessary analog amplifiers. The prototype of this system was, in fact, developed for the self-adjustment research. A self-administered consonant-contrast test (CCT) is under development for use in this research. During formative evaluation of the CCT, the processing platform was used to measure the effects of low-pass filtering in listeners with normal hearing (a crude simulation of hearing loss). Because the UCSD platform provides two sets of filters whose center frequencies differ by half an octave, it was possible to measure with cut-off frequencies from 177 to 8000 Hz in half-octave steps. Group mean composite contrast score, after correction for chance, fell to around 50% at a (6 dB) cut-off frequency of around 700 Hz. Under this condition, however, scores for consonant place and voicing were 20% and 90%, respectively. A similar pattern was found for listeners with sensorineural hearing loss during the self-adjustment research. These two examples of application involve four types of user: speech/hearing scientists, hearing-aid researchers, audiologists, and listeners with hearing loss. Each has specific needs beyond those of the computer scientist who will be accessing the open-source code. The design and development of accessible, transparent, and modifiable interfaces to meet the needs and capabilities of differing user-types could be critical to the successful application of open-source speech-processing platforms to the improvement of hearing health-care. [Work funded by NIDCD grants R33DC015046 and R01DC015436.]

**B3-P-03: Discrimination of Frequency-Gain Curve Adjustments**

Benjamin Caswell-Midwinter\(^1\) and William M Whitmer\(^{1,2}\)

\(^{1}\) MRC  
\(^{2}\) CSO Institute of Hearing Research - Scottish Section

The frequency-gain curve (FGC) is the fundamental hearing-aid parameter. FGC adjustments in the clinic, whether matching to real-ear targets or patient feedback, are routine. Furthermore, preference and self-fitting paradigms often involve direct paired-comparisons of FGCs. Fittings could be inefficient and patient feedback unreliable if adjustments are less than what is discriminable. To examine what FGC adjustments are discriminable, we measured the just-noticeable differences (JNDs) for frequency-band level increments from stimuli processed with a prescription FGC.

JNDs were measured with hearing-impaired participants using a fixed-level, same-different procedure. Thirty-eight participants discriminated pairs of speech-shaped noises (SSNs), and forty-one participants discriminated pairs of male, single-talker sentences. For SSNs, single-band increments were made to a 0.25 kHz low pass (LP) band, 0.5-4 kHz octave bands, and a 6 kHz high-pass (HP) octave band. For sentences, single-band increments were made to 0.25, 1 and 4 kHz bands, and multi-band increments were made to LP, band-pass (BP) and HP bands. Broad-
B3-P-04: Data-driven auditory profiling as a tool for defining Better hEAring Rehabilitation (BEAR)

Raul Sanchez-Lopez1, Michal Fereczkowski1, Federica Bianchi1, Sébastien Santurette2, and Torsten Dau1
1 Technical University of Denmark
2 Department of Otorhinolaryngology, Head and Neck Surgery and Audiology, Rigshospitalet

Background While the audiogram still stands as the main tool for selecting hearing-aid compensation strategies in audiological clinics, there is ample evidence that loss of hearing sensitivity cannot fully account for common difficulties encountered by people with sensorineural hearing loss, such as understanding speech in noisy environments. Forty years after R. Plomp proposed his attenuation-distortion model of hearing impairment, it remains a challenge to address the distortion component, mainly related to suprathreshold deficits, via adequate clinical diagnostics and corresponding hearing-aid compensation strategies. Inspired by the different auditory profiling approaches used in the literature, a major aim of the Better hEAring Rehabilitation (BEAR) project is to define a new clinical profiling tool, a test battery, for individualized hearing loss characterization. Methods The proposed BEAR approach is based on the hypothesis that any listener’s hearing can be characterized along two dimensions reflecting largely independent types of perceptual distortions. In order to keep the approach as neutral as possible, no a priori assumption was made about the nature of the two distortion types. Instead, a statistical analysis method, combining unsupervised and supervised learning, was applied to existing data. The aim was to provide a tool to help define the two distortion types, such that potentially relevant tests for classifying listeners into different auditory profiles could be identified. So far, the data from two auditory profiling studies were reanalyzed based on this approach. First, an unsupervised-learning technique including archetypal analysis was used to identify extreme patterns in the data, forming the basis for different auditory profiles. Next, a decision tree was determined to classify the listeners into one of the profiles. Results The data-driven analysis provided consistent evidence for the existence of two independent sources of distortion, and thus different auditory profiles, in the data. The results suggested that the first distortion type was related to loss of sensitivity at high frequencies as well as reduced peripheral compression and frequency selectivity, while the second distortion type was linked to binaural temporal-fine-structure processing abilities as well as low-frequency sensitivity loss. The audiogram was not found to reflect an independent dimension on its own, and the most informative predictors for profile identification beyond the audiogram were related to temporal processing, binaural processing, compressive peripheral nonlinearity, and speech-in-noise perception. The current approach can be used to analyze other existing data sets and may help define an optimal test battery to achieve efficient clinical auditory profiling.

B3-P-05: Psychoacoustic factors influencing speech perception in noise and reverberation for individuals with and without hearing impairment

Blair Ellis1, Ewan Macpherson1, Paula Folkeard1, Chris Allen1, Susan Scollie2, and Prudence Allan1
1 Western University - National Center for Audiology
2 University of Western Ontario

For people with hearing loss, complex listening environments such as reverberant and noisy spaces may cause listening penalties that can negatively influence speech perception, yet the sources of individual variability underlying such listening penalties remain somewhat elusive. For example, we may be able to
The current project aims to examine listening performance of two groups of older adults who have either normal hearing (n=18; mean age = 69.1 years, REPTA: 17.6 dB HL) or clinically diagnosed hearing loss (n=15; mean age = 75.8 years; RE-PTA: 39.1 dB HL). All participants were assessed for suprathreshold hearing abilities. This evaluation included measures of auditory processing abilities such as frequency resolution, frequency discrimination, temporal resolution, binaural hearing, and clinical measures of hearing ability. Additionally, listening performance in noise and/or reverberation was measured using the Hearing in Noise Test (HINT); listeners with hearing loss wore hearing aids fitted to the DSLv5-adult target for this task. The HINT task was completed under nine conditions using three environments (sound booth, sound booth with non-spatial simulated reverberation, and reverberant room), in the contexts of quiet, co-located noise, or spatially separated noise.

We obtained a typology of hearing performance profiles across all 33 participants by conducting principal components analysis on the scores from HINT across all nine conditions. Individual principal component scores were then predicted using hearing thresholds, age, and suprathreshold measures of hearing ability. Preliminary results from ongoing analyses suggest that most of the variation between individuals occurs along the continuum of normal hearing to impaired hearing. Although hearing threshold data accounts for variability on this dimension, orthogonal within-group variability was not explained by the same threshold measures. Instead, within-group variability was best explained by measures of temporal and frequency resolution, as well as a measure of binaural hearing ability.

**Results:** Preliminary analysis of the results from the first 40 listeners is presented. The analysis includes distributions of estimated parameters (in particular, related to compression rate). Additionally, the influence of training on performance in the notched-noise task is analyzed and will be considered during development of a mobile-app for auditory filter estimation.

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**B3-P-06: ECHO – estimating cochlear compression in the clinic**

*Michał Fereczkowski, Sébastien Santurette, and Ewen MacDonald*

**Technical University of Denmark**

**Background:** Currently, sensorineural hearing-loss diagnosis and hearing-aid fitting mainly rely on the audiogram. However, individuals with similar audiograms may differ to a large extent in their suprathreshold (and aided) performance. While researchers and clinicians agree that supra-threshold measures are necessary to better describe individual loss, few if any such measures have been adopted in clinical practice. Residual, cochlear compression is a prime example. Long test times, large measurement variability, and lack of clear application guidelines are major obstacles preventing clinical application of compression measures. This is unfortunate in light of research showing that residual compression is not correlated with the audiogram for mild-to-moderate hearing losses and that it may be a significant predictor of performance in supra-threshold tasks, including speech perception in noise. The aim of the Echo project is to evaluate clinical feasibility of two measures of residual compression – fixed-frequency DPOAEs and auditory-filter shape estimates. An additional aim is to investigate into potential dependencies between cochlear, hearing-aid compression and aided performance, on a large group of listeners. Method The listeners are recruited from a hospital clinic in Copenhagen and participate in four test sessions, with an overall duration of 6-7.5 hours. Five different experiments are performed on the better ear. During separate sessions, DPOAE growth functions (obtained with three swept-tone and fixed-frequency paradigms), auditory-filter shapes, and temporal masking curves are estimated at 1 and 2 kHz. The listeners undergo short guided-stepwise training before each of the psychoacoustic experiments. Additionally, each listener fills in the SSQ12 questionnaire about their performance with the current hearing-instruments and compression characteristics of the instrument from the tested ear are measured in the Affinity test box.

**B3-P-07: Fit-to-target and SII normative data for DSL v5.0 adult fittings**

*Paula Folkheard*, *Hasan Saleh*, *Danielle Glista*, and *Susan Scollie*

1. National Centre for Audiology - Western University
2. Western University
3. University of Western Ontario

**Background:** Currently, sensorineural hearing-loss diagnosis and hearing-aid fitting mainly rely on the
Previous work by Moodie et al (2017) examined fit to target and Speech Intelligibility Index (SII) values for pediatric fittings using the Desired Sensation Level version 5.0 (DSL v5.0) child targets. These data have supported the development of normative ranges on clinical evaluation tools as well as cross-project comparisons. Due to prescription differences in gain and output between pediatric and adult targets, these normative ranges are not applicable to adult fittings. The purpose of this study is to measure the range of deviation from DSL v5.0 targets for adult fittings, and to generate normative ranges for aided SII for these fittings. Modern hearing aids fitted to DSL v5.0 targets to two hundred and ten (210) adult ears will be reviewed in a retrospective chart review. These hearing aids were fitted between 2016 and 2018. All fittings were verified using probe microphone measurements with the Audioscan Verifit VF2. Data from three styles of hearing aids: behind-the-ear; in-the-ear, and receiver-in-the-canal hearing aids from four manufacturers are available for analysis. Deviation from target will be calculated using root mean squared error (RMSE) from target by input level, following methods recommended by McCreery et al. (2013). Speech Intelligibility (SII) scores for each fitting will be extracted to obtain normative values for these fittings across a wide range of input levels. Effects of factors such as style of hearing aid, degree of hearing loss, frequency and level will be examined. We will also examine the impact of including a broad versus narrow frequency range within the fit to targets deviation, and how this interacts with degree of hearing loss. Results will be used to develop fit to target and SII typical performance data for adult fittings to implement into clinical instrumentation. Comparison to previously published literature and various systems for appraising fit to target deviations will be provided.

**B3-P-08: Improving pediatric real-ear-to-coupler difference predictions and aided audibility using immittance**

Anastasia Grindle\(^1\), Elizabeth Walker\(^2\), Meredith Spratford\(^3\), King Chung\(^4\), Marc Brennan\(^5\), and Ryan McCreery\(^6\)

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Collection of a real-ear-to-coupler difference (RECD) is an essential part of hearing aid verification in children, ensuring safe amplification and appropriate audibility for speech. When measurement of an RECD is not possible, age-related average RECD values may be used. Significant variability in RECD values exist among children of the same age (+/- 10-15 dB). This variability limits the accuracy of hearing aid verification with age-average RECD. If variability in RECD is related to differences in ear acoustics, the use of tympanometry may lead to improved accuracy of predicted RECD values. The purpose of this study was to examine whether a measure of outer and middle ear status, 226 Hz tympanometry or wideband tympanometry, may be used to create improved predictions of RECD in children, as compared to age-average RECD. Additionally, this study examined whether predicted RECD, including immittance data, influenced hearing aid fittings and speech intelligibility. In experiment 1, 226 Hz tympanometry ear canal volume and compliance was collected from 266 children. Using a linear mixed model, we analyzed the extent to which inclusion of immittance information into age-average RECD predicted a child’s measured RECD. In experiment 2, we measured wideband tympanometry and RECD in 149 children. We analyzed the extent to which inclusion of ear canal volume, absorbance, and age-average RECD predicted a child’s measured RECD via linear mixed modeling to create the immittance-predicted RECD. In experiment 3, we performed simulated real-ear speechmapping verification using three different audiograms. The hearing aid was programmed to Desired Sensation Level (DSL) child version 5.0 prescriptive targets, using five different percentiles of measured, age-average, and immittance-predicted RECD, representing the large range of RECD values seen in children of the same age. Root-mean-square (RMS) error from the measured RECD DSL targets and average level (65 dB SPL) aided speech intelligibility index (SII) were calculated for each condition. Our analysis indicated that inclusion of ear canal volume and immittance measures improved predictions of a child’s measured RECD, with the magnitude of improvement varying across frequency. In hearing aid fittings, the RMS error from DSL target differed significantly based on RECD type for the three audiograms, with immittance-predicted RECD resulting in a hearing aid output closer to prescriptive targets generated using measured RECD than age-average RECD. These findings support the development of a tool that would allow audiologists to utilize immittance to more accurately predict RECD values when measured RECD values cannot be obtained.
B3-P-09: An evaluation method for simultaneous assessment of ability to match target gain, sound quality, and feedback performance in hearing aids

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Evaluation of feedback performance in hearing aids is not a trivial task. Typically, the hearing aid design has to be a compromise between the ability to match target gain, sound quality, and the feedback performance; sacrificing the performance in one or more aspects would typically allow better performance in the remaining aspect(s). Hence, a comprehensive feedback evaluation should cover all aspects simultaneously.

We present a novel evaluation method that considers all three aspects. More specifically, in order to evaluate the ability to match target gain, we fit each test hearing aid on a manikin based on a well-known prescriptive target and a very steeply sloping hearing loss (to provoke feedback), e.g., the NAL-NL2 target and the so-called standard audiogram S2. We then conduct real-ear measurements to compare the insertion gains provided by test hearing aids to the reference insertion gain defined by the prescriptive target. Furthermore, in a MUSHRA test setup, we evaluate sound quality in static situations using lab recordings from each test hearing aid. Mixed speech and music signals are used to obtain a detailed sound quality evaluation. Finally, we apply an exploratory blind test to evaluate feedback performance. Each test person rates own perceived feedback annoyance level on each test hearing aid in several critical dynamic feedback situations, such as covering the ear/hearing aid with a phone/hand etc.

In a test with 21 normal hearing participants, we used this method to compare the premium products from six international leading hearing aid manufacturers A-F to a prototype hearing aid including an updated feedback system. We present the comparison results of these six commercial hearing aids and our prototype to demonstrate that our evaluation method provides a balanced and comprehensive view of the ability to match target gain, sound quality, and the feedback performance. We also observed, interestingly but maybe not surprisingly, that the manufacturers A-F prioritized these three somehow contradicting aspects very differently in their products.

Furthermore, we relate the presented method to another often-used evaluation method, which consists of determining the max insertion gain before feedback and sound quality evaluation; the main differences compared to the presented method are that an expert examiner determines the presence of feedback, and the sound quality is evaluated based on normalized hearing aid output signals.

B3-P-10: Masking effects on perceptual cues in hearing-impaired ears

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Objectives: Current hearing-aids mainly work by improving audibility. However, the most common complaint is "I can hear the speech but I can't understand it." This has led us to investigate the strategy of the hearing-impaired (HI) ear in listening to speech. In this study, effects of varying masking on primary cues in HI ears are analyzed, with and without the presence of conflicting cues (Li and Allen). Using an error pattern analysis, the results provide insights into the HI ear's listening strategy.

Design: Two consonant-vowel (CV) identification experiments were conducted on five normal-hearing (NH) and ten HI subjects. Four plosive consonants (tokens) /t,k,d,g/ paired with vowel /a/, in CV context, were used as target stimuli. The CVs were presented at signal-to-noise ratios (SNRs) of 0, 9, and 18 dB. In experiment one the primary cue for each CV was modified in 3 ways: removed, attenuated, and amplified. Experiment two was similar to experiment one except the conflicting cues were removed. Subjects' responses were analyzed based on 1) consonant errors and 2) confusion patterns (token confusions vs SNR).

Conclusions: The analysis shows that HI listeners are using the same primary cue as NH ears. The strength of the primary cue is a critical quality for low-error HI speech perception, especially in the presence of noise. The analysis suggests that conflicting cues are the dominant sources of error for both NH and HI, especially when noise masks or signal modification removes primary cue.

The HI subjects were divided into two groups, based on their average error: 1) a low error group (LEG) which contains six subjects and 2) a high error group (HEG) with four subjects.

Overall, the LEG showed results similar to the average normal hearing (ANH) group, for both experiments, but with higher error. The removal of the conflicting cues did not have notable impact on their performance. However, the subjects in the HEG exhibit
sensitive reaction towards the presence of conflicting cues. Depending on the subject from the HEG, the removal of conflicting cues can enhance or diminish speech perception. Interestingly, a few ears from HEG depend on conflicting cues for correct recognition. We shall explore this observation further.

Our results suggest that the LEG are more likely to benefit from hearing aids. Thus, it would be clinically useful to classify subjects in this way to predict success with their aided condition, as well as to improve the fitting algorithm.

**B3-P-11: Using the Vanderbilt Fatigue Scale to explore the effects of hearing loss and device use on listening-related fatigue in adults and children with hearing loss**

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Severe, recurrent fatigue is a common complaint of individuals suffering from chronic health conditions and can have significant negative effects on quality of life. Evidence from generic fatigue measures suggests that adults (AHL) and children (CHL) with hearing loss are also at increased risk for long-term, listening-related fatigue. However, there are no measures designed specifically to assess listening-related fatigue. Such measures are essential for improving our understanding of, and developing interventions to reduce, listening-related fatigue and its consequences. To address this need we continue to refine a package of patient-reported outcome measures designed to reliably assess listening-related fatigue— the Vanderbilt Fatigue Scales for adults (VFS-AHL) and children (VFS-CHL) with hearing loss. To obtain a more complete picture of listening-related fatigue in children the VFS-CHL has child, parent and teacher versions. The scales were developed using rigorous, qualitative methods. Focus groups and cognitive interviews were used to gather rich qualitative data from AHL and CHL, as well as the children’s parents and teachers. These data were transcribed, coded, and used to create a multi-dimensional construct map reflecting relevant domains and behavioral indices of fatigue severity. We have collected data from ~600 adults using a preliminary, 60-item version of the VFS-AHL. More than 600 adults also completed a 10-item version of the VFS-AHL during a validation study. Similarly, we have data from 170 children, 319 parents, and 250 teachers/service providers using a preliminary, 250 teachers/service providers using a preliminary 60-item version of the VFS-CHL. Item Response Theory (IRT) will be used to examine these preliminary data from the VFS-CHL scales and a subset of high quality items will be selected to create final versions for validation. In addition to VFS data, we also collected demographic (e.g., age, gender) and audiologic information (e.g., self-reported degree of hearing loss, hearing device use) from all AHL and CHL respondents. In this presentation we use the data set described above to examine associations between ratings of listening-related fatigue and 1) degree of self-reported hearing loss and 2) use of hearing aids or other auditory prosthetics (e.g., cochlear implants). Preliminary analyses of VFS-AHL validation data, revealed a systematic increase in listening-related fatigue as degree of self-reported hearing loss increased up to a severe range. Interestingly, fatigue ratings decline significantly when self-reported losses increase to a profound range. Analyses of the effects of hearing loss on our child data, as a function of respondent type (child, parent, teacher) are ongoing and will be discussed.

**B3-P-12: Comparison of unaided and aided psychoacoustic tuning curves with normal hearing and hearing impaired listeners**

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Listeners with sensorineural hearing loss usually exhibit poorer frequency selectivity than normal listeners, even when their elevated thresholds are compensated by amplification. In the current study, we examine the role of linear insertion gain in restoring the hearing impaired (HI) listener’s frequency selectivity by measuring the psychoacoustic tuning curves (PTC’s) with and without amplification. The system used in the study integrates the fast PTC test (Sek et al. 2011) and a real time MATLAB program which applies an insertion gain, prescribed using the NAL-R formula (Byrne et al. 1986) based on the participant’s hearing thresholds, to the stimuli.

The hearing threshold levels of both normal hearing (NH) and HI participants were first measured with the probe frequency at 250 Hz, 500 Hz, 1000 Hz, and 2000 Hz, before the fast PTC measurements. For HI listeners, insertion gain computed as a function of frequency based on the hearing threshold levels was applied to the stimuli while measuring the ‘aided’ PTC’s. For NH listeners, a gain of 10 dBLSL was applied to the stimuli instead while obtaining the equivalent ‘aided’ PTC’s.
Recent studies have further revealed that individual preference towards increase directionality with respect to SNR is bounded by the speech reception threshold (SRT) and the acceptable noise level (ANL). However, SRT and ANL estimates vary for different listeners and also for different environmental conditions. Consequently, predicting individualized preferences based on those individual estimates remains very challenging in everyday listening conditions. The main goal of our study is to examine the relevant individual factors such as subjective attributes, personality traits and hearing loss and their ability to predict the strength of directionality preferred by the listener in selected real-life situations.

B3-P-14: Skills transference validity of a probe tube placement training simulator

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Probe tube placement is an important aspect of real-ear verification which must be performed successfully for a proper hearing aid fitting. Unfortunately, numerous studies have found large numbers of poor fittings along with an absence of verification in some clinics, citing lack of confidence and lack of training as possible explanations. To facilitate training in probe microphone placement and verification, a training simulator was developed that allows trainees to practice probe placement while receiving feedback. An anthropomorphically correct 3D printed mannequin head/neck/shoulders was created with soft silicone ears based on a bank of clinical CT scans. The mannequin head and silicone ear were coupled with a mounted camera system that analyzes the location of any objects being inserted into the mannequin’s ear. Software was developed that provides the user with real-time feedback regarding the location of the probe, and quantitative metrics such as probe-to-ear-drum distance and time to insert. A practice mode encourages students to practice their probe tube placement in a pre-clinical scenario without the need for a partner/volunteer, whereas a test mode can be used by instructors to evaluate users’ performance. While a previous validation study evaluated the realism and training ability of the simulator with experts, the current project will determine whether trainee usage of

Our preliminary results showed that PTC’s measured with HI participants using aided stimuli were higher in Q10 value (narrower or sharper tuning curves) than those using unaided stimuli, which is not observed in the NH participants’ results with a gain of 10 dB SL. Furthermore, the tip frequencies of the HI PTC’s were drifted further away from the probe frequency.

B3-P-13: Towards individualized automatic control of directionality strength in binaural hearing aids

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In recent years, with the constant development in semiconductor technology, a new wave of hearing aids with improved directional technology was introduced into the market. High-end hearing instruments are now equipped with a wireless link capable of transmitting high-speed bidirectional audio data from ear-to-ear. This means that each hearing instrument has additionally access to the microphone signal received from the instrument on the other side of the head. Therefore, having simultaneously access to all microphone signals unleashed a series of new signal processing algorithms such as binaural beamforming, which can provide directionality with a narrower beamwidth towards the front direction. As a result, recent studies have also shown that these new types of hearing aids can provide an even more efficient solution to speech understanding in background noise. However, it is also clear that this mode of high directionality should only be applied when necessary mostly in demanding listening situations. Thus, hearing aid manufacturers provide a so-called ‘automatic mode’ programmed in the hearing instrument. When this mode is enabled, the strength of directionality of the hearing instruments is automatically adjusted by various estimators sensing the environment such as signal-to-noise ratio (SNR), background noise level and speech activity level estimated from the microphone signals. For instance, as the background noise level increases, directionality gradually increases from a wide (Omni-directional) to a narrower beam-width. However, this scheme does not necessarily always result in individual satisfaction or preference. For example, for the same environment and at the same background noise level or SNR, different users might prefer more or even less directionality.
this simulator results in clinical skills directly trans-
ferable to clinical scenarios.

Twenty-five novice clinicians with less than two
hours of experience with probe tube placement par-
ticipated in this study. A randomized controlled trial
was completed in which participants were split into
one of two groups: the simulator group, and the con-
trol group. Each group underwent a pre-test, two-
week training period, and a post-test. Participants in
the control groups used traditional methods of train-
ing (with a partner/volunteer) while participants in the
simulator group supplemented their traditional train-
ing by using the simulator. Pre- and post-tests con-
sisted of each participant performing a probe tube
placement on a volunteer where a self-evaluation was
completed, an expert evaluated their technique, and
the trainees measured the real ear unaided response
(REUR) and the real ear to coupler difference
(RECD) on the human volunteer. Results from the
two different groups will be analyzed to determine
any effects of pre- or post-testing, effects of simulator
use, or interactions between these.

B3-P-15: Towards a clinically viable spectro-temporal modulation test

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Hearing-aid processing is typically adapted to the
needs of the individual listener by means of the pure-
tone audiogram, which captures some important as-
pects of hearing impairment in terms of sensitivity
loss. However, the crucial ability of a listener to un-
derstand speech in adverse conditions may be
strongly influenced by supra-threshold processing
deficits and is thus not necessarily well-represented
by the audiogram. Recent studies (e.g., Bernstein et
al., 2013; Bernstein et al., 2016) have suggested that
a spectro-temporal modulation (STM) test might be
well-suited to capture supra-threshold deficits in
hearing-impaired (HI) listeners, demonstrating that
STM detection performance was related to the per-
formance in speech-in-noise tests. Bernstein et al. (2013)
showed a strong link between STM thresholds and
speech intelligibility, whereas Bernstein et al. (2016)
demonstrated a somewhat weaker connection be-
tween the two and additionally reported that no STM

B3-P-16: Real-world achieved gains and subjective outcomes of a wide-bandwidth contact hearing aid fitted with CAM2

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Two large studies were conducted under FDA and
IRB oversight on a novel light-driven contact hearing
aid (the Earlens), which directly drives the eardrum to
achieve functional audibility from 125-10,000 Hz
(Gantz et al, 2017). These studies provide the first
large-scale evaluations of real-world experience with a
non-simulated hearing device capable of imple-
menting the CAM2 algorithm, which prescribes gain
for frequencies up to 10 kHz, with nearly 60 dB of
low-level insertion gain prescribed for a hearing loss
of 80 dB HL at 10 kHz. The studies were conducted
primarily for regulatory and commercial reasons.
However, analysis after the fact allowed us to examine the prescribed and achieved gains as verified using functional gain measures, and to examine subjective outcomes.

Data were obtained for 94 participants across eight clinical sites. Participants were fitted with the Earlens system. Initial fittings used CAM2-prescribed gains for experienced users, and adjustments were made when required according to participant preferences for loudness and comfort or at the audiologist’s discretion. Participants wore the devices for a period of at least 90-120 days. Outcome measures were: prescribed versus adjusted output and gain; frequency-specific functional gain; self-perceived benefit assessed with the Abbreviated Profile of Hearing Aid Benefit (APHAB); and a custom questionnaire. Self-perceived benefit results were compared to those for unaided listening and to ratings with participants’ own as-fit acoustic hearing aids.

The prescribed low-level gain from 6 to 10 kHz averaged 51.5 dB across all ears. After adjustment, this decreased to 44 dB. Measured functional gain averaged 39 dB from 6 to 10 kHz. Ranges and comparisons between programmed and achieved gain will be discussed. The APHAB communication scores revealed a significant improvement over unaided listening, averaging 27 to 33 percentage points across the three sub-scales. Most participants preferred slightly less gain at 8-10 kHz than prescribed for experienced users by CAM2, preferring similar gains to those prescribed for inexperienced users. Gains at high frequencies were very high relative to those for current popular prescriptive algorithms typically available and used in air conduction hearing aids. These studies indicate that participants not only tolerate but also benefit from the high gains prescribed by CAM2 and achieved using the Earlens system in the real world.

Reference:
Loss of function of inner hair cells, primary auditory neurons and synapses between inner hair cells and neurons can all lead to a reduced flow of information in the auditory nerve and to less precise coding of the properties of suprathreshold sounds. Several researchers have measured amplitude-modulation (AM) detection thresholds to try to detect such loss of function. However, AM detection thresholds may not be sufficiently sensitive to reduced fidelity in the auditory coding of AM. Also, outer hair cell dysfunction can improve the ability to detect AM, probably because of the loss of cochlear compression, and, when present, this may offset deleterious effects of less precise coding. The “Envelope Regularity Discrimination” (ERD) test was designed to overcome these problems. In one randomly chosen interval of the two-alternative forced-choice task, 8-Hz sinusoidal AM of a sinusoidal carrier is present. In the other interval, the AM has the same mean rate and depth but is irregular in rate and amount. The task is to pick the interval in which the AM sounds irregular. The amount of irregularity is specified by the “irregularity index” (II), which is varied from trial to trial using an adaptive procedure to estimate the value leading to 70.7% correct. If the presentation of envelopes in the auditory system is “noisy,” then regular AM may sound somewhat irregular. This should lead to worse performance on the ERD test. Synaptopathy and neuropathy may selectively affect auditory neurons with high thresholds. Hence, our initial data were gathered using a signal level of 80 dB SPL. A broadband threshold-equalizing noise was used to limit the range of characteristic frequencies of the neurons that respond to the signal. The ERD test involves stimuli with clearly audible AM depths. Loudness recruitment may magnify the perceived AM depth, but this effect will be similar for the two intervals of the forced-choice trial and should have little or no influence on performance of the task. To confirm that the baseline modulation depth had little effect, performance was measured for normal-hearing subjects using a carrier frequency of 4 kHz and clearly-audible modulation depths ranging from 0.2 to 0.5. The effect of modulation depth was very small. However, marked individual differences were
found. The individual differences were explored using a baseline modulation depth of 0.3. Although practice effects were observed, marked differences persisted with extended practice. The origin of the individual differences is being explored.

8:20 AM  **C3-O-5: Subjective evaluation of different attack times in single microphone noise reduction**

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Single microphone noise reduction (NR) is one of many features in modern hearing aids (HA) with the purpose to reduce unwanted background noise and works by adjustments of the frequency specific hearing aid gain. Previous research has shown beneficial effects of NR in HAs in terms of listener preference. Those studies were primarily focused on static properties of NR such as the amount of gain reduction that is preferred. This research focuses on dynamical aspects of NR by investigating the effects of time constants in a NR algorithm on subjective preference. Time constants in NR are often embedded deep in the algorithm making them undesirable to use in a research set-up as they can interact in a complex manner. Therefore, we tried to isolate the effects of time constants within NR by delaying the gain of a minima-controlled recursive NR algorithm by means of temporal exponential smoothing. In this way, we introduced a single attack time to determine which speed of changes listeners prefer. The attack time was varied with time constants between 0 and 200 ms. 16 normal hearing and 16 hearing impaired subjects participated in three listening tests to determine speech intelligibility, discriminability and subjective preferences of our test signals. We verified that all signals had only marginal effects on speech intelligibility and that the differences were detectable. The subjective preference results show that NH subjects – on average – have a significant preference for temporally smoothed NR with attack times of 100 ms or 200 ms. HI listeners prefer noise-reduced signals over unprocessed signals but we found no significant effect of attack time on subjective preferences on a group level. The inter-individual variability is large, which makes it reasonable to assume that attack times can have a significant effect on subjective preferences for some individuals. Preferences for NR settings might be governed by a tradeoff between noise reduction and signal distortion. Future research will explore this individual tradeoff for dynamic settings in NR as well as other NR features such as the preferred amount of gain reduction, in order to optimize hearing aid fitting.

8:40 AM  **C3-O-6: Preferred hearing aid gain settings for music-listening using a 3D modified simplex procedure implemented with the Open Source Master Hearing Aid platform**

Jonathan Vaisberg, Steve Beaulac, Danielle Glista, Maaike Van Eeckhoutte, Ewan Macpherson and Susan Scollie

National Centre for Audiology; Western University

Hearing aids can improve the audibility and intelligibility of speech, and are often used with prescriptive gain formulæ to personalize the gain and output. However, the majority of hearing aid prescriptions have been developed for use with speech, and are often evaluated using measures of speech intelligibility, which may not be appropriate for understanding sound quality impacts of prescribed gain and output for non-speech stimuli such as music. This study seeks to determine whether a preferred gain setting exists which differs between speech and music, using overall preference as a criterion across normal-hearing and/or hearing-impaired listeners. We hypothesize that preferred gain settings will diverge from prescribed settings more for music-listening than for speech-listening.
This study uses the Open Source Master Hearing Aid (openMHA, Herzke, et al, 2017), a hearing aid simulator, which consists of algorithms found in a basic hearing aid processing chain. OpenMHA settings are easily adjustable using programming software, facilitating real-time adjustment of parameters for hearing aid research studies.

In this study, openMHA provides individualized fittings with easily-adjustable gains. We have paired the openMHA with a MATLAB-automated modified simplex procedure (MSP). This MSP is a multivariate paired-comparison procedure that evaluates preferences between stimuli differing in two or more electroacoustic parameters. MSP has historically been used to assess stimuli differing in the amount of low- and high-frequency gain provided. In the present study, MSP systematically adjusts three parameters: gains in low-, mid-, and high-frequency bands. Participants initially listen to a stimulus (speech or music) under their prescribed gains. Through a series of stepping rules, MSP systematically adjusts the openMHA gains within a three-dimensional space and in real-time. MSP terminates once the participant no longer prefers further adjustments, at which point the most recent adjustment is defined as their preferred setting.

This study is currently in data collection with normal-hearing listeners. Prescribed gains are calculated using the DSL v5.0 method with dBHL thresholds and individual ear acoustics as inputs. Pilot data reveal that more gain is preferred in the low- and high-frequency bands, and less gain is preferred in the mid-frequency band. Listeners tend to increase low and high frequencies more for music than for speech, and decrease mid frequencies more for speech than for music. Preliminary results suggest a need for the development of music-based prescriptions and show a benefit with personal fine-tuning. Data analysis is ongoing, with next steps including testing with listeners who have hearing losses.

9:00 AM C3-O-7: Perceptual strategies for consonant discrimination in individuals with and without hearing loss

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¹ École Normale Supérieure  
² Starkey

Understanding what acoustic cues human listeners rely on to discriminate speech sounds in noise, how they use such cues, and how these perceptual strategies are impacted by hearing loss, is an important step toward designing more effective speech-processing algorithms for hearing-impaired or normal-hearing individuals.

In this study, we investigated these questions with a psychophysical approach based on earlier work using psychophysical reverse correlation to uncover perceptually relevant acoustic cues for consonant discrimination. In this approach, recordings of natural speech (here, vowel-liquid-plosive-vowel) signals are altered by the addition of two noise “blobs”. The times and frequencies of these blobs are selected so that they are likely to enhance, or degrade, certain spectrotemporal regions of the signal, some of which were identified (in earlier experiments using the reverse correlation method) as corresponding to acoustic cues used by listeners for discriminating the plosive consonant; one cue (primary) corresponds approximately to the transition of the second and third formants, while the other (secondary) corresponds approximately to the first formant. By systematically, and independently, varying the blob-to-signal ratio for these two blobs, we could infer the locations, and the relative importance (or perceptual weight), of these perceptually relevant acoustic cues, for a given listener.

We used this approach in three groups of listeners: listeners with a normal audiogram (‘normal-hearing’); listeners with high-frequency sloping hearing loss; and listeners with an approximately flat, or shallowly sloping, audiogram. We hypothesized that the perceptual importance of the higher-frequency (second- and third-formant transition) cue would be reduced, relative to that of
the lower-frequency (first-formant) cue in listeners with high-frequency sloping hearing loss, compared to listeners with normal-hearing, and listeners with ‘flat’ hearing loss, even after the amplification of frequency-dependent amplification to restore high-frequency audibility.

The results turned out to be only partly consistent with our hypothesis, with some listeners following the predicted patterns while others clearly did not. The substantial interindividual variability in cue-weighting strategies among normal-hearing and hearing-impaired individuals, which is apparent in the data, suggests a need for individually-tailored speech-in-noise processing in these two populations, if more effective speech-enhancement in noise is to be achieved.

In addition, the modified psychophysical reverse-correlation technique developed for this study leverages prior knowledge to speed-up the assessment of individual perceptual cues/strategies for consonant perception, and could be easier to apply in clinical populations than the — more time-consuming — alternatives proposed in earlier studies.

9:20 AM  C3-O-8: Development of an integrated Repeat and Recall Test (RRT)

Christopher Slugocki, Francis Kuk, and Petri Korhonen
Widex ORCA

Poor working memory is thought to contribute to deficits with speech-in-noise processing yet few audiology-based working memory measures exist for clinical use. To this end, we developed and assessed the validity of an integrated Repeat and Recall Test (RRT) for measuring speech-in-noise recognition, working memory capacity, listening effort, and tolerable time in adults with and without a hearing loss. The RRT was administered to 20 normal hearing adults (mean age = 50.1 years). Listeners were tested in quiet and at signal-to-noise ratios (SNRs) of -5 through +15 dB, in 5 dB steps, using 5 sets of speech materials that provided either high or low degrees of semantic context. Performance-intensity functions were compared among lists to verify the equivalence of speech materials. The RRT was also administered to a group of 16 hearing-impaired adults (mean age = 64.1 years) in quiet and at SNRs of 0 through +15 dB. Performance-intensity functions from hearing-impaired listeners were compared to those of the normal hearing group. Repeat and recall scores were assessed for test-retest reliability and compared with the Hearing in Noise Test (HINT) and the Reading Span Test (RST), and with subjective measures of listening effort and tolerable time. The potential effect of semantic context on these outcome measures in normal and hearing-impaired listeners was also examined. All 5 sets of speech materials yielded similar repeat and recall performances in normal hearing listeners. In both normal and hearing-impaired listeners, repeat performance reflected expected performance deficits at degraded SNRs and low context speech materials. Moreover, 50% speech reception thresholds (SRT-50) derived from repeat performance curves were significantly predictive of performance on the HINT. Listener recall performance in quiet and in noise was significantly related to performance on the RST. Ratings of listening effort reflected listener performance on both repeat and recall portions of the RRT, whereas reports of tolerable time were mainly related to repeat performance. Both groups of listeners were dependent on semantic context cues. Our results demonstrate the viability of the RRT in assessing a listener’s speech-in-noise recognition and verbal working memory capacity, and her/his concurrent subjective perception at different SNR conditions for speech materials that are either high or low in contextual cues.
C3-P-17: Hearing-aid self-adjustment by experienced hearing-aid users using the Goldilocks protocol
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The purpose of the work is to develop a protocol for user self-adjustment of hearing aids and to determine its efficacy and candidacy. The UCSD Open-source Speech-processing Platform was used to provide amplification using real-time, six-band, processing of microphone input from ear-level transducer assemblies. An Android tablet was used to control the amplification via the Goldilocks software which allows user adjustment of overall volume, high-frequency boost, and low-frequency cut.

Twenty experienced hearing aid users with mild to moderately-severe hearing loss participated. Participants adjusted each controller in a fixed order while listening to continuous speech until they were satisfied. Adjustments were made in quiet or in background noise (signal-to-noise ratio = +6 dB). All adjustments started from a generic frequency response based on the NAL-NL2 prescription for a mild-moderate sloping hearing loss. To examine the effect of exposure to amplification between the first and second adjustment, 10 of the participants took a speech-perception test after an initial self-adjustment; the other 10 completed a non-auditory game. Following user self-adjustment, speech recognition testing was completed in both quiet and noise to examine differences between scores for the generic starting response and the self-adjusted response.

Mean real-ear aided responses for the user-adjustments 1) were higher than the generic starting response and 2) were at or above the NAL-NL2 prescription for both the noise and quiet conditions. Unlike our previous study using pre-processed stimuli, there was no evidence that exposure to the intervening speech recognition test resulted in higher mean output for the second adjustment. Individual preferences were, however, closer to the NAL target after the second adjustment – suggesting the need for a minimum of two adjustments. Speech recognition scores for the final self-adjusted responses were significantly higher than for the generic starting response.

Conclusions:
1) Participants were able to adjust amplification to responses close to the NAL-NL2 prescription
2) There is no evidence (for mild to moderately-severe losses) that an individualized prescription-based starting response is required for adequate self-adjusted audibility
3) At least two adjustments are needed.

C3-P-18: Factors influencing enjoyment of music with hearing aids
Rémi Marchand1,2,3, Jörg Buchholz1,2,3, Harvey Dillon1,2,3, and Valerie Looi4
1 The HEARing CRC
2 Macquarie University
3 National Acoustic Laboratories
4 Sydney Cochlear Implant Centre

Currently, hearing aids (HAs) are adjusted to compensate for individual hearing loss primarily to maximize the clarity and comfort of speech. Electroacoustic characteristics and settings of HAs may be ideal for speech recognition, but not for music enjoyment. Some aspects of the signal processing involved in HA design may interfere with the enjoyment of music. As an alternative to the standard fitting methods, most of the manufacturers offer different processing programs for customers in need of a specific amplification for music. However, recent studies suggested that these music programs may not improve significantly
the experience of music listening and can still be optimized.

**Aims:** To better understand the musical listening habits of HA users, to identify the main issues they experience while listening to music and to develop signal processing recommendations specifically for music.

**Methods:** A survey was conducted, consisting of 42 multiple-choice and open-ended questions. 151 respondents were recruited from two large databases available at the National Acoustic Laboratories, providing detailed information about the respondents such as their age, type and degree of hearing loss, and information about their HAs. Currently at its initial stage, a follow-up study involves controlled listening experiments to further understand the signal processing strategies preferred by participants when listening to music in relation to the problems highlighted by the survey. Manipulations of the signal processing comprise changes in compression ratio and frequency specific amplification.

**Results:** The survey showed that HA users listen mainly to recorded music at home and use the HAs in their universal program. The most prevalent problems identified are difficulties in understanding lyrics, the soft passages of music being too soft, lack of clarity of the music, poor tonal quality, the music being too shrill and the music being too loud. Results suggest that compression algorithms should be improved. In the listening experiment, the preferred amount of compression applied by the HAs is expected to be influenced by the dynamic range of the music stimuli at the input of the HAs while its frequency spectrum is expected to influence the preferred frequency-specific amplification provided by the HAs.

**Conclusions:** About 30% of the users are dissatisfied with the performance of their HAs while listening to music and experience problems that need to be addressed. Future research should derive amplification prescription schemes specifically targeted at optimizing the enjoyment of music with HAs.

**C3-P-19:** Comparisons of criteria for prescriptive target proximity in children who wear hearing aids

Ryan McCreery, Elizabeth Walker, Meredith Spratford, and Marc Brennan

1 Boys Town National Research Hospital
2 University of Iowa
3 University of Nebraska-Lincoln

Children with higher aided audibility through their hearing aids have stronger language development than peers with poorer aided audibility, even after accounting for the child’s degree of hearing loss. An important determinant of audibility is the proximity of the hearing aid output to prescriptive targets. To date, a criterion of a root-mean-square (RMS) error < 5 dB based on the difference between prescriptive target and hearing aid output at 500 Hz, 1000 Hz, 2000 Hz, and 4000 Hz has been suggested for children. The goal of this analysis was to determine if using a more stringent criterion of 3 dB RMS error led to improvements in audibility after controlling for degree of hearing loss. Hearing aid verification data for 309 children with mild-to-severe hearing loss at 1,214 study visits were analyzed to examine differences in aided audibility for soft (50 dB SPL) and average (65 dB SPL) input levels for children with < 3 dB RMS error compared to children with 3-5 dB RMS error or > 5 dB RMS error. The child’s age and degree of hearing loss were included as covariates in the analysis. Children with < 3 dB RMS error from prescriptive targets had better aided audibility than peers with > 3 dB RMS error after controlling for age and degree of hearing loss. This effect was larger for soft speech input levels and children with greater degrees of hearing loss. This analysis suggests a 3 dB RMS error criterion may result in higher aided audibility for speech for children who wear hearing aids than the 5 dB RMS error criterion that is currently recommended.

**C3-P-20:** Aided sentence perception as related to the identification of syllable constituents and the use of context

James Miller, Charles Watson, Meredith Marjorie Leek, David Wark, Pamela Souza, Sandra Gordon-Salant, Jayne Ahlstrom, and Judy Dubno

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Hearing-aids users’ identifications of 109 syllable constituents (45 onsets, 28 nuclei and 36 codas) and words in sentences are studied. The participants have PTAs less than 75 dB HL, have used aids for at least three months prior to their participation, and are without complicating conditions. It was found for 59 participants that aided scores in quiet ranged from 23% to 85% correct identifications. Also, it was found that
the identification of these syllable constituents (phonetic details) in noise is highly correlated with their identification scores in quiet (Miller et al. 2017, JASA, 141(4), 2933). Similar results are now found for 113 aid users. This implies that the aided listener’s problem in noise is not due to the noise, per se, but rather is due to reduced ability to identify syllable constituents, whether in quiet or in noise. This, in turn, implies that identifiability of syllable constituents (phonetic details) should be a major focus for hearing-aid design and fitting. New data (submitted for publication) is presented that demonstrate that an aided listener’s sentence perception in noise is well predicted by that listener’s syllable-constituent identification when combined with that listener’s use of context. The syllable constituent, contextual model of speech perception allows estimation of each aided listener’s use of context. It is shown that when syllable-constituent scores are above about 55%, sentence scores are relatively high and individual differences are relatively small. However, as syllable-constituent scores fall below 55%, listeners with better context usage have much higher sentence scores than those with poorer context usage. It is recommended that hearing-aid designers and fitters should minimally strive to achieve syllable-constituent scores of about 71% correct in quiet as that provides syllable-constituent scores of about 55% correct in noise. This, in turn, is predictive of high sentence scores in noise, no matter the listener’s context usage. The data and analyses supporting these conclusions will be presented. In summary, these findings can be interpreted to support the notion that aided speech perception does not depend solely on the audiability of peaks in the speech spectrum, but rather depends, more directly, on the identifiability of phonetic details.

C3-P-21: Hearing aid experience and background noise affect the robust relationship between working memory and speech reception in noise
Elaine Ng¹ and Jerker Rönnberg²
¹ Oticon A/S
² Linköping University

The robust relationship between speech reception in noise and working memory in listeners has been well documented in the literature. The aim of this study is to examine how background noise (4-talker babble and stationary noise) and duration of hearing aid use affect the robust relationship between working memory and speech reception. One hundred and forty-eight participants with at least 2 years of hearing aid experience in both ears and symmetrical mild to moderate sensorineural hearing loss were included in the data analyses. Working memory was assessed using the reading span test, semantic word-pair test and visual-spatial working memory test. Matrix sentences in Swedish (Hagerman sentences) were used to measure speech reception in noise. Results showed that the 4-talker babble noise condition yielded a stronger overall correlation between working memory and speech reception performance than in a stationary noise background. The participants were further divided into three groups based on their hearing aid experience (2 to 5 years, 5 to 10 years and over 10 years). The correlations in the stationary noise was significantly weaker in the least experienced group than the most experienced group. In 4-talker babble, however, there was no significant difference between the strength of correlations among the three hearing aid experience groups. This study revealed that more explicit processing of working memory is deployed when listening in a multi-talker babble background than in a stationary noise. This study also suggested that in stationary noise, the matching processes (c.f. Ease of Language Understanding model; Rönnberg et al., 2008, 2013) were more efficient for long-term than relatively less experienced users when perceiving speech.

C3-P-22: The effect of carrier bandwidth and intensity on amplified spectral modulation detection thresholds
Kristi Oeding and Evelyn Davies-Venn
University of Minnesota

Listeners with hearing loss must often listen to amplified speech at high intensity levels to restore lost audibility. Several studies have shown that at high intensities minimal differences exist between individuals with normal compared to those with impaired cochlear function. The loss of the compressive non-linearity of outer hair cell gain at high intensities results in a passive cochlear response that broadens auditory filter bandwidths for all listeners. Most of these studies, however, have used narrowband measures such as notched-noise masking to quantify intensity-induced degradation in spectral processing. A few recent studies have used spectrally modulated rippled noise as a fast and accurate method of quantifying broadband spectral processing in naïve and skilled listeners. This measure of broadband spectral processing has also shown strong correlation with clinical speech tasks that are often used to quantify hearing aid outcomes. Our previous work compared inten-
sity level effects on individuals with normal compared to impaired cochlear function using a fixed sound pressure level. Although our findings showed differences between the two groups, unequal audibility at fixed sound pressure levels was a likely confound that may have obscured some of the true differences in broadband spectral processing for individuals with normal versus impaired cochlear function. This study assessed the relationship between carrier bandwidth and intensity on spectral processing using amplified signals with individualized frequency gain responses using a hearing aid simulator.

A within-subject repeated measure design was used to measure spectral-ripple modulation detection (SMD) thresholds using a spectrally modulated noise signal with varying carrier bandwidths from 1 octave to 4 octaves and intensity levels from 60 to 90 dB SPL. A three forced-alternative choice paradigm was used with each participant deciding which of the three intervals contained the modulated signal. This task required minimal training and each threshold took less than 5 minutes. Results to date have shown expected trends of degradation in spectral processing with increased signal intensity. Audibility controlled SMD thresholds will be reported for each intensity and bandwidth condition for individuals with hearing loss.

C3-P-23: An app for validation of hearing aid fitting with speech and noise presented in three sound channels
Jon Øygarden
Norwegian University of Science and Technology

The European standard EN 15927 ‘Services offered by hearing aid professionals’ clause 5.5.4 proposes three main methods for verification of improvement in hearing ability (A. speech audiometry in sound field, B. surveys with real-ear measurements and C. questionnaire concerning the perceived benefit from the hearing aid system). The standard states that at least one of these methods should be used and the results reviewed with the client.

To make possible validation with method A an app developed in Matlab is available for Windows PCs utilizing the short sentences (three words) in the Norwegian MATRIX test. Three loudspeakers are mounted in a horizontal line on the wall. The listener sits 1 meter in front of the center loudspeaker, which always delivers the speech material. The right and left speakers are mounted with 1 meter distance from the center speaker, making it possible to deliver noise from ±45°.

The app is designed to measure speech reception thresholds (SRT) by an adaptive method in two sessions, first without hearing aids and then with the hearing aids. In each session three measurements will be performed: 1 – without noise, 2 – with noise front and finally 3 – with noise from both sides. The difference in SRT between measurement 2 and 3 gives spatial release from masking (SRM).

To maximize the SRM the selected noise is two instances of MATRIX sentences played backwards, giving a mean SRM of 9.7 dB with standard deviation 1.6 dB for a group of young normal hearing listeners. The recommended noise level is 45 dB SPL.

The app has two different graphic user interfaces (GUI). The audiologist-GUI where the audiologist scores number of words correctly recognized or the user-GUI where the listeners marks results on a 3x10 matrix of all the words. A bootstrap analysis during the measurement gives an estimate of the standard error of the result.

Results will be presented from monaural and binaural measurements on a group of young normal hearing listeners. Audiograms and results from some hearing-impaired individuals with and without hearing aids are discussed.

C3-P-24: Comparing self-adjusted amplification and gain preferences across listening contexts
Trevor Perry¹, Peggy Nelson¹, and Dianne Van Tasell²
¹ University of Minnesota
² EarMachine

New technology can allow listeners’ real-time self-adjustment of custom amplification parameters to accommodate preferences in changing listening conditions. Previous results (Nelson et al, in review; Perry et al, in review) indicate that large inter-subject gain differences are observed in self-adjusted studies. Listeners with mild-to-moderate hearing loss adjusted a real-time simulation of a multichannel compression hearing aid, with gain/compression parameters adjustable via a simple user interface (EarMachine). A subset of listeners tended to set gain higher than their NAL-NL2 based prescriptions for most speech-in-noise conditions; other listeners tended to choose less gain than prescribed by NAL across several noisy speech conditions. Differences in gain changes were unexplained by age, hearing loss, or previous hearing aid use. Further, the gain adjustments had little effect.
The ability to recognize emotion provides a foundation for social relationships by facilitating identification of communicative intent—a skill necessary for successful interpersonal interactions. Social relationships can directly impact happiness (Bertara, 2005), success (Martin, 2009), and even physical health (House, 1988). It follows, then, that disruption of emotional interpretation or recognition can significantly impact quality of life.

A study by Picou (2016) investigated subjective ratings of emotional valence (pleasant vs. unpleasant) of hearing impaired listeners for everyday sounds. Individuals with hearing loss showed a reduced emotional range, meaning that these individuals, on average, experienced emotions that were less extreme, particularly positive emotions, compared to normal-hearing listeners. This range was further constricted when the stimuli were presented at 80 dB instead of 50 dB. In addition to experiencing an impaired range of emotions, listeners with hearing loss also experienced a negative shift in valence (i.e., perceived unpleasantness) when listening to amplified sounds.

Goy et al. (2016) found that older individuals with hearing loss had significant difficulty in identifying emotion as conveyed by a talker, compared to both younger and older adults with normal hearing.

The results of these aforementioned studies demonstrate some underlying difference between normal-hearing and hearing-impaired listeners on these emotional constructs. Further, and extremely important for hearing healthcare providers, is the impact of hearing aids on the domains of emotion recognition (recognizing the emotion of a communicative partner) and emotional responses to non-speech stimuli.

The current study at the Phonak Audiology Research Center (PARC) was designed to assess the impact of custom hearing aid amplification on the domains of a) emotion recognition and b) emotional ratings of "pleasantness" in response to non-speech stimuli. It was of interest whether hearing aid amplification improved the ability to recognize emotion as conveyed by a communication partner, or further degraded performance in this domain. It was also of interest how the introduction of hearing aid amplification would impact the emotional range of valence ratings in response to non-speech sounds.

The results showed that the introduction of hearing aid amplification neither helped nor hindered emotional recognition ability. The current study also showed that hearing aid amplification reduced the average ratings of valence in response to non-speech sounds. It is important to further investigate the impact of hearing aids on these emotional domains.

**C3-P-25: The effect of hearing aid amplification on emotion recognition and emotional responses**

*Lori Rakita*  
*Phonak LLC*

The ability to recognize emotion provides a foundation for social relationships by facilitating identification of communicative intent—a skill necessary for successful interpersonal interactions. Social relationships can directly impact happiness (Bertara, 2005), success (Martin, 2009), and even physical health (House, 1988). It follows, then, that disruption of emotional interpretation or recognition can significantly impact quality of life.

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**C3-P-26: 10+ years of Acceptable Noise Level Test: What have we learned?**

*Karrie Recker*  
*Starkey*

It’s been over a decade since the influential paper by Nabelek et al. (2006) was published that claimed that the Acceptable Noise Level (ANL) test could predict, with 85% accuracy, who would be successful with hearing aids. The ANL test measures one’s most comfortable listening level (MCL) for speech, and the highest background noise level (BNL) that one is “willing to put up with” without becoming tense or tired while listening to and following the speech. ANL = MCL – BNL, in dB. Those who are willing to tolerate high levels of background noise (i.e., those with ANLs < 7 dB) are likely to be successful with hearing aids, while those who are intolerant of background noise (i.e., those with ANLs > 13 dB) are unlikely to be successful with hearing aids. Success was defined by these researchers as, “the individuals wore their hearing aids whenever they needed them.”

This Nabelek et al. paper spurred much conversation regarding the definition of hearing aid “success;” and dozens of papers have since been published using the ANL test as researchers have rushed to understand its predictive ability, its repeatability and how its results are impacted by various test variables, patient factors and hearing aid features. Along the way, a handful of studies have attempted to determine the perceptual criterion by which individuals choose their ANLs.
An overview of the ANL test and a summary of what we’ve learned since the introduction of this test will be provided. Recommendations will be made regarding the future use of this test in research and the clinic.

**C3-P-27: Subjective judgments on frequency-gain curve adjustments**

Benjamin Caswell-Midwinter¹ and William M. Whitmer²

¹ MRC
² MRC/CSO Institute of Hearing Research-Scottish Section

Fine-tuning is routine in hearing-aid fittings. While the efficacy of the fine-tuning process is inconclusive, widely used expert systems exist which establish common troubleshooting descriptors and solutions which are focused on adjusting the frequency-gain curve (FGC); for example, a solution for the descriptor “tinny” is to “decrease high-frequency gain” (Jenstad et al., 2003; Thielemans et al., 2017). These expert systems however, are not perceptually validated, nor operationalised. Fitting-assistants based on these systems vary across manufacturers, and treat audiometric profiles homogeneously. To examine the relationship between FGC adjustments and common descriptors, we assessed thirty-three hearing-impaired participants’ judgments on adjusted FGCs. Paired-comparison judgments were made between reference and alternate FGCs using male, single-talker sentences. Alternate FGCs were adjusted from prescribed reference FGCs. Adjustments were made across a ± 12 dB range in 4 dB steps separately at low (0.25 and 0.5 kHz bands), mid (1 and 2 kHz bands) and high (4 and 6 kHz bands) frequencies. Participants judged whether a second sentence sounded “better”, “worse”, or “no different” (ND) to a first, explicating their preference by choosing a descriptor judgment from a closed-set of descriptors. Participants compared each alternate FGC to a reference a total of 10 times. Stimuli were presented over headphones. Across participants, descriptor judgments broadly corroborated troubleshooting guides, although the FGC adjustments necessary to elicit these descriptors were large. Furthermore, there was wide inter-participant heterogeneity. Linear interpolations suggested 50% thresholds for ND judgments of ± 5 dB for low adjustments, ± 6 dB for mid adjustments, and ± 9 dB for high adjustments. The amount of ND judgments per participant was positively correlated with their hearing threshold. These results suggest that troubleshooting with small adjustments may not be noticeable by listeners, and that the individual variation in descriptor meaning poses challenges to automated troubleshooting.

**Reference:**


**C3-P-28: Auditory tests for characterizing individual hearing deficits: The BEAR test battery**

Raul Sanchez-Lopez¹, Michal Fereczkowski¹, Federica Bianchi², Mouhamed El-Haj-Alif³, Tobias Neher², Torsten Dau¹, and Sébastien Santurette²

¹ Hearing Systems Group, Technical University of Denmark
² Institute of Clinical Research, University of Southern Denmark

The Better hEAring Rehabilitation (BEAR) project seeks to develop and assess new clinically feasible strategies for individualized hearing-loss diagnosis and hearing-aid fitting. The aim is to improve current clinical practice, where the fitting process relies on the pure-tone audiogram and trial-and-error methods. These usually result in inconsistent practices and patient dissatisfaction and inefficient service. Existing evidence suggests that the audiogram does not sufficiently describe supra-threshold performance of hearing-impaired listeners. Detailed characterization of hearing deficits can be complex. Therefore, one aim of the BEAR project is to design a hearing test battery for classification of listeners into a small number of auditory profiles. If successful, this BEAR test battery may be refined and reduced to form the basis for improved profile-based hearing-aid fitting protocols.

**Method**

Based on the reanalysis of existing auditory profiling data and on criteria of their feasibility, time efficiency, and evidence from the literature, eleven potential tests for inclusion in a clinical test battery were selected. The proposed tests were divided into six categories: audibility, middle-ear analysis, speech perception, binaural-processing abilities, loudness perception, and spectro-temporal resolution. Thirty hearing-impaired listeners with symmetric mild to severe sensorineural hearing loss were selected from a clinical population of hearing-aid users. All listeners performed every test included in the battery. The participants were tested in a clinical environment and did not receive systematic training on any of the tasks. Results The considered tests have so far shown poten-
tial for auditory profiling. The analysis of the preliminary results will focus on the ability of each test to pinpoint individual differences among the participants, interrelations among the tests, as well as their usability for the target clinical population. Importantly, a parallel study will evaluate the extent to which the outcomes of these tests can be used for hearing-aid fitting. Finally, the current test battery will be refined for implementation in clinical practice, based on the results of a data-driven analysis for auditory profiling.

C3-P-29: Estimation of auditory filter shapes across frequencies using machine learning
Josef Schlittenlacher, Richard E. Turner, and Brian C. J. Moore
University of Cambridge

When fitting a hearing aid, the level-dependent gain prescribed at each frequency is usually based on the hearing loss at that frequency. This often results in reasonable fittings for a typical cochlear hearing loss, but may fail when the individual frequency selectivity and/or loudness growth are different from what would be typical for that hearing loss. Individualised fitting based on measures of frequency selectivity might be useful in improving a fitting, for example by reducing across-channel masking. A popular measure of frequency selectivity is the notched-noise method, but this test is time-consuming. To reduce testing time, Shen and Richards (2013) proposed an efficient machine-learning test that determines the slope of the skirts of the auditory filter (p), its minimum response for wide notches (r), and detection efficiency (K). However, their test did not determine asymmetries in the auditory filter, which are important to consider during fitting to reduce across-channel masking. The test proposed here provides a time-efficient way of estimating the auditory filter shape and asymmetry as a function of center frequency. The noise level required for threshold is estimated for a tone with frequency fs presented at 15 dB SL in nine symmetric or asymmetric notched noises with notch edge frequencies between 0.6 and 1.4 fs. Using only narrow to medium notch widths provides good information about the tip of the auditory filter, which is of most importance in determining across-channel masking for speech-like signals (but the tail is not well defined). The nine thresholds for a given fs can be used to fit an auditory filter model with three parameters: the slopes of the lower and upper sides (pl, pu) and K. In practice, these model parameters are estimated as a continuous function of fs, and fs is varied across trials over the range 0.5-4 kHz. The stimulus parameters on a given trial (fs, notch condition, noise level) are chosen to maximally reduce the uncertainty in the model parameters, exploiting the covariance between thresholds for adjacent values of fs. Six subjects have been tested so far. The whole procedure took about 45 minutes per ear. The lower slopes typically corresponded with values expected from the audiogram and a cochlear hearing loss. The upper slopes were steeper in some cases, although not necessarily across the whole frequency range.

Reference:

C3-P-30: Evaluation of noise reduction systems with subjective listening effort measurements
Michael Schulte1, Volker Kühnel2, Markus Meisl3, Kirsten C. Wagener4, and Melanie Krueger5
1 Hoerzentrum Oldenburg
2 Sonova AG

Hearing impaired people often complain about effortful listening in noisy environments. To reduce the perceived listening effort (LE) and to increase the acceptance of hearing aids, noise reduction systems were developed. The question is how to evaluate and proof the benefit of single channel and directional noise reduction (NR) systems. Especially for single channel NR algorithms the effect on speech intelligibility is little but there might be an effect on LE. A handy method to evaluate the NR systems of hearing aids is the adaptive categorical listening effort scaling (ACALES, Krueger et al, 2017). In ACALES the signal-to-noise ratio (SNR) is varied adaptively to adjust the individual SNR range from “no effort” to “extreme effort” using a 14-step scale. There are two different ways to set the SNRs in ACALES: first, the standard way that keeps the noise presentation level constant and adaptively varies the speech presentation level (adaptive level method) and an alternative way that varies the SNR but keeps the overall level of noise and target constant (constant level method). In the latter case the rating is not influenced by the overall presentation level. A study was conducted in which these two methods were used and compared with speech test results. We were interested in the test-retest reliability and the sensitivity for single channel NR. 15 hearing impaired subjects participated. LE measurements and speech tests in noise were performed with speech from 0° direction and
noise from 135° (S0N135) for the directional NR system and S0N0 for the single channel NR system. There was a clear significant benefit of the directional NR system for speech intelligibility and LE. For the single channel NR system there was no effect in intelligibility but in some subjects rated LE to be less with NR on (not significant) with the adaptive level method. With the constant level method, the LE difference between NR on and off was statistically significant. The test-retest reliability was high for the adaptive level method but worse for the constant level method which might be due to different training effects. To improve the reliability the training effects for this constant level method will be further evaluated. All in all both methods are easy to perform and have pros and cons which will be discussed in the poster.

Reference:

C3-P-31: A pre-post intervention study of hearing aid amplification: Results of the Emotional Communication in Hearing Questionnaire (EMO-CHEQ)

Gurjit Singh1, Melanie Kreuger2, Jana Besser3, Lars Wietoska4, Benno Wagner4, Stefan Launer5, and Marcus Meis2
1 Phonak Canada
2 Hörzentrum Oldenburg GmbH
3 Sonova AG
4 Vitakustik

The objective of the study was to evaluate a self-report questionnaire, the Emotional Communication in Hearing Questionnaire (EMO-CHEQ, see Singh et al., in press, Ear and Hearing). The EMO-CHEQ is designed to assess experiences of hearing and handicap when listening to speech that contains vocal emotion information. In several steps, the EMO-CHEQ questionnaire was translated into German by a forward-backward procedure. In a previous study conducted in Germany, cross-sectional data collected using an ex-post-facto design revealed only slight effects of hearing aid amplification. In the current follow-up study, we evaluated the Unitron Moxi Fit hearing aid before and after provision with experienced hearing aid users (EXU) and first-time users (FTU). N=158 end-users (N=88 FTU, mean age=68.0 yrs./PTA better ear=32.1 dB HL and N=70 EXU, mean age=71.4 yrs./PTA better ear=47.4 dB HL) of Vitakustik clinics across Germany were recruited. The research measures included a speech-in-noise test, and a short questionnaire battery with items from the SSQ, IOI-HA, the full German version of the EMO-CHEQ (see contribution, Meis et al., IHCON 2018), and items assessing listening effort. On the EMO-CHEQ, for all the sub-scales (“Speaker characteristics”, “Speech production”, “Situational Hearing”, and “Socio-emotional well-being”), a significant improvement of HA amplification from 0.5 – 1.0 scale units on a scale ranging from 1-5 (p<0.015 to p<0.001) was demonstrated by means of non-parametrical repeated analyses. Additionally, listeners benefitted significantly from hearing aid provision on the speech-in-noise measure and the SSQ. Overall, the pattern of results suggests that Moxi hearing aids are associated with observable benefits on both subjective and objective outcome measures, and that the EMO-CHEQ questionnaire may be a sensitive outcome measure to assess experiences of hearing and handicap when listening to speech that contains vocal emotion information. Implications of this work will be discussed.

C3-P-32: The effect of extended bandwidth on loudness and DSL v5.0 targets

Maaike Van Eeckhoutte, Paula Folkeard, Danielle Glista, and Susan Scollie
National Centre for Audiology, University of Western Ontario

The Desired Sensation Level (DSL) algorithm can be used to prescribe gain or output across frequencies and input levels, when fitting hearing aids to children and adults. The prescription depends on the level, individual hearing thresholds, age-appropriate ear canal values, and the nature of the hearing (pediatric versus acquired). The current version, DSL v5.0, delivers targets for a frequency range between 200 and 8000 Hz, and prescribes targets for each ear independently. However, those with asymmetrical hearing losses may require different listening levels per ear. Also, modern hearing aids offer output bandwidths that may extend above 8000 Hz, and some hearing aid prescriptions and/or verification systems have begun to offer targets and measurement in the 10,000 to 16,000 Hz range. This extended bandwidth is not accounted for by the DSL targets that were developed for hearing aids with a more limited bandwidth. The purpose of this study was to investigate the perceived aided loudness growth of sounds when wearing hearing aids with a limited versus an extended bandwidth. Additionally, we investigated if loudness is balanced between the ears at DSLv5.0 targets when wearing hearing aids with an extended bandwidth. Adults were tested in the free field. The stimuli were: (a) high-pass
filtered pink noise with a cut-off frequency of 1000 Hz for testing of binaural loudness balance; and (b) spoken sentences of the Connected Speech Test for testing loudness growth. All participants were fitted binaurally with extended-bandwidth hearing aids, and verified to meet targets from the DSL v5.0 prescription using clinical verification procedures. The limited-bandwidth condition was created by reducing the output at high frequencies in the fitting software. Loudness growth was measured using the Contour test. Loudness balance across the ears was measured using an Up and Down track adjustment procedure, keeping the better ear at a fixed gain and providing a variable gain for the worse ear. Preliminary results indicate that sounds are perceived louder for the extended-bandwidth condition compared to the limited-bandwidth condition across all input levels, when wearing hearing aids fitted to DSL v5.0 targets. Furthermore, on average, balanced loudness was found at 2.5 dB below the DSL v5.0 target for the extended-bandwidth condition. Further analyses and data collection will support investigation of factors such as degree of hearing loss and age group of adults versus children, and may suggest modifications to the DSL v5.0 approach.

Friday, August 17

SESSION C4. Models and Their Application to Hearing Aids

Session Chair: Arthur Boothroyd

11:10 AM  C4-O-1: Auditory processing models and their potential application in hearing technology

Torsten Dau
Hearing Systems Group, Technical University of Denmark

Auditory processing models provide a powerful framework to both represent and interpret the results from a variety of experiments and to further understand the functioning of different parts of the auditory system. An important category of functional computational models seeks to capture the essential signal transformations along the auditory pathway, rather than their specific physiological substrates, and helps generate hypotheses that can be quantitatively tested for complex systems at different functional levels. These models can also help determine how a deficit in one or more functional components affects the overall operation of the system. The first part of this presentation describes some current trends in quantitative modeling of speech perception in challenging acoustic conditions, inspired by coding principles from physiology at periphery and mid-brain stages of processing. While such models can account reasonably well for speech intelligibility data from normal-hearing listeners, the prediction of data from individual hearing-impaired or aided listeners remains challenging. The second part considers compensation strategies in hearing instruments inspired by auditory models. Current compensation schemes, such as dynamic range compression, loudness compensation or speech enhancement, aim at processing the signal such that the perception of the aided signal in the hearing-impaired listener matches the perception of a normal-hearing listener. While some approaches are promising, nonlinear system compensation is generally difficult to achieve in real-life situations and real-time applications. Moreover, even for the case of a purely peripheral impairment, consequences at more central stages can be manifold and complex and are typically less well understood. Current compensation strategies aim to restore peripheral processing but the restoration of cues at central stages may be equally relevant to consider. This, in turn, requires models that accurately capture such higher-level processing. Some modeling perspectives are finally described that attempt to bridge this gap between peripheral/mid-brain and central processing using artificial neural network architectures. Such networks are optimized to solve real-world auditory tasks, such as speech recognition, and are currently matching
the performance of human listeners. The degree to which task-optimized models can be viewed as models of the ‘real’ biological system are discussed. Overall, it is argued that the primary relevance of computational auditory signal-processing models remains the description of the transformations of the acoustical input signal into its essential ‘internal’ representations. More detailed descriptions of this process may, in turn, allow for more sophisticated hearing-aid compensation strategies to be developed.

11:50 AM  C4-O-2: Overcoming the quagmire of hearing loss heterogeneity: Towards optimization of hearing aids via distinct genetic hearing loss populations

Ian Bruce¹, Susan Stanton², Anne Griffin³, Amanda Morgan², Matthew Lucas², Jill Lowther³, and Terry-Lynn Young³
¹ McMaster University
² Western University
³ Memorial University Newfoundland

The development of optimal hearing aid amplification strategies is greatly hindered by the significant heterogeneity in etiologies and diversity of hearing loss pathologies. One tactic to deal with such diversity would be to subdivide the general hearing-impaired population based on an extensive battery of auditory assessments, however it is not clear what subdivisions are appropriate beyond the standard groupings according to etiology, and in many cases even the etiology is unclear.

Our group has developed an alternative genomics-based approach, leveraging the existence of several extended families in Newfoundland, Canada, that have inherited distinct forms of genetic hearing loss from a historical founder member of a Newfoundland community. Each of these families has a distinctive pattern of auditory pathology due to a specific genetic mutation, such that homogeneous auditory deficits are apparent among affected members within a family, while being different across families. These include a KCNQ4 mutation leading to progressive high frequency hearing loss, a WFS1 mutation producing low-frequency hearing loss, and a FOXL1 mutation giving rise to otosclerosis.

This study involves the deep phenotyping of members of each of these families, including the acquisition of advanced electrophysiological recordings (ABR and ECochG), word perception in quiet and noise, psychophysical tuning curves, wideband tympanometry, and DP OAE growth functions, in addition to routine audiometric measures. The data from these recordings will be used, along with animal models of some of these gene mutations, to inform the incorporation of appropriate pathology into the Bruce et al. (Hear. Res. 2018) computational model of the auditory periphery. Quantitative predictions of the electrophysiological and speech intelligibility data will be used for model validation. In this presentation, we will report on the analysis of data collected from several members of each family and on preliminary modeling results. The impaired models will subsequently be used to optimize hearing aid amplification strategies to compensate for the specific deficits caused by the different genetic mutations. We also anticipate that these optimized algorithms will provide improved benefit for hearing aid users with different etiologies but who have similar patterns of pathology, thus greatly extending the utility of this optimization approach.

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C4-P-01: Evaluating the impact of methodology on calculating signal modification resulting from hearing aid processing

Melinda Anderson¹, Varsha Rallapalli², James Kates³, Pamela Souza², and Kathryn Arehart⁴
¹ University of Colorado School of Medicine
² Northwestern University
³ University of Colorado, Boulder

The ability to analyze the acoustic output of a hearing aid allows researchers to study how hearing-aid processing modifies a signal. In recent work, we have characterized the amount of signal modification resulting from commercial hearing aid processing by using a cepstral correlation metric. This metric quantifies the change in the temporal and spectral modulation of the hearing-aid processed signal compared to a reference signal. These measures have been completed by making recordings of hearing aid output using an anechoic test box in order to isolate the device for recordings. However, the use of the anechoic test box does not allow for consideration of the effect of the coupling technique used to fit a commercial hearing aid. There are advantages to making recordings using a method able to accurately capture the output of a clinically fit hearing aid using the listener's coupling technique. A second method of creating hearing aid recordings using an acoustic manikin has been shown to be feasible in our laboratory, which allows for modifying the coupling technique. However, this methodology results in differences in the metric value given the lack of device isolation. The goal of this study is to quantify the effect of different recording methodologies on signal modification calculations (i.e. cepstral correlations) when the input signal and hearing aid feature settings are kept constant and the coupling system and recording methodology are varied. Results to date indicate similar calculated values when the target input speech signal is presented in background noise, with a greater difference between methodologies when the input speech signal is in quiet. The results will be presented in the context of the acoustic impact of alterations to the coupling technique and recording methodology on cepstral correlation calculations. The outcome of this study may influence how hearing aid researchers consider hearing aid recordings for the purpose of acoustic analyses. [Portions of this work were funded by NIH R01 DC012289 (Souza and Arehart)]

C4-P-02: Determining the insertion gain of hearing aids by confusing-syllables identification

Jing Chen, Hongying Yang, and Xihong Wu
Peking University

With the development of internet and smart phones, it became possible to implement the basic function of hearing aids on smartphones, e.g. multi-channel compression, to help hearing-impaired listeners improve their speech perception. The insertion gain for each frequency band can be initially determined by the listener's audiogram. However, it is difficult to get an audiogram by testing on a smart phone due to the rigorous requirement of pure-tone audiometry. Hence, in this work we aimed to propose a method for determining the insertion gain by confusing-syllables identification. The basic assumption was that the loudness perception was correlated with some auditory features of speech identification, e.g. the critical feature for identifying easily-confusing word-pairs. First, we constructed a speech corpus including all easily-confusing word-pairs with Mandarin Chinese. The words were monosyllabic or disyllabic. For the disyllabic words, only one syllable was easily confused and the other syllable was the same within a word pair. The discriminative auditory feature for each word-pair was analyzed based on the loudness model (Moore and Glasberg, 2004). To select the typical word pairs, the identification of which is highly correlated to their auditory features, a speech identifica-
A perception experiment was conducted with 18 normal hearing listeners. In the experiment, all candidate word-pairs were processed by a hearing-impaired model, in which the loudness loss was simulated for each of six frequency bands. According to the experimental result, 34 word-pairs were selected and composed the final speech corpus, in which the identification performance was highly correlated to the loudness based auditory feature ($r = 0.8$, $p < 0.001$). Secondly, a neural network was used to model the mapping between word-pairs’ identification and the two parameter values of the loudness model. These parameters could be used to calculate the specific loudness slope, and further determined the insertion gain of each frequency band. The input of the neural network included the identification result and the 6-dementional auditory feature, which was the specific loudness level difference of each of the six frequency bands for a given word-pair. The output was the evaluation of the two parameter values. With this neural-network based model, a listener’s insertion gain could be predicted by the confusing-syllables identification. During the speech test, an adaptive procedure was used to select the word pairs. The experiment result indicated that the predicted insertion gain was comparable to the insertion gain calculated with audiogram.

C4-P-03: Perceptual evaluation of signal-to-noise-ratio aware wide-dynamic range compression
Tobias May, Borys Kowalewski, and Torsten Dau
Hearing Systems Group, Technical University of Denmark

Fast-acting wide-dynamic range compression (WDRC) can improve speech audibility and the ability to listening-in-the-dips in hearing-impaired listeners. However, WDRC can simultaneously introduce temporal envelope distortions that reduce speech modulations and may disrupt the natural foreground-background relationship of speech in the presence of noise. Recently, May et al. (2018) proposed a novel hearing-aid compensation strategy, which utilizes information about the local signal-to-noise ratio (SNR) in individual time-frequency (T-F) units to select compression time constants. The system applies a short release time for speech-dominated T-F units while the processing is linearized by a long release time for noise-dominated T-F units. This SNR-aware compression scheme achieves a similar effective compression of speech components as conventional fast-acting WDRC while avoiding distortion such as amplification of noise in speech pauses. This contribution evaluates the subjective benefit of SNR-aware compression in comparison to conventional fast-acting and slow-acting WDRC by measuring speech intelligibility and quality over a wide range of SNRs in a group of hearing-impaired listeners. The results will help assess the applicability of this novel compensation strategy in hearing aids.

References:

C4-P-04: Hearing aid sound quality assessment and analysis using paired comparisons
Martin McKinney, Krishna Rodemerk, Mo Movahedi, and Tao Zhang
Starkey Hearing Technologies

The subjective sound quality of hearing aids can be complex and difficult to characterize, which makes it onerous to specifically address. In this study, we established a method to measure subjective sound quality and to link those measures to specific acoustic and signal characteristics. We used a paired comparison paradigm to measure the subjective sound quality of hearing aids from three different manufacturers in which participants indicated their overall preference. Four listening conditions were tested: soft speech in quiet; loud speech in quiet; music; and loud speech in babble noise with a +4 dB SNR. Normal-hearing (N=15) and hearing-impaired (N=15) participants were fitted with each manufacturer’s default/proprietary fitting formula. Normal-hearing participants were fitted using the averaged (moderately sloping) audiogram from our participant database. Stimuli for the study were prepared by playing the raw/clean stimuli over loudspeakers to hearing aids positioned on a KEMAR mannequin, recording from the KEMAR in-ear microphones, and subsequently correcting each participants real-ear-to-KEMAR difference. Prepared stimuli were presented to participants through in-ear monitors for the paired-comparison study. Preference data were analyzed and modeled in an effort to identify principal dimensions of preference. Subject-specific and objective acoustic factors were assessed in terms of their contribution to overall preference.

C4-P-05: Quantifying the range of signal modification in clinically-fit hearing aids
Varsha Rallapalli, Melinda Anderson, James Kates, Synn Sirow, Kathryn Arehart, and Pamela Souza
Northwestern University
Hearing aids provide various signal processing techniques with a range of parameters to improve the listening experience for a hearing-impaired individual. In recent work, we reported significant differences in signal modification for mild versus strong signal processing in commercially available hearing aids. In this study, we extend this work to clinically prescribed hearing aid fittings based on standard-of-care guidelines. The goals of this project are to determine the distribution of signal modification in clinically fit hearing aids, and the effects of acoustic factors such as signal-to-noise ratio (SNR) and input level on these signal modifications. We report a comprehensive set of measurements for hearing aid fittings that were obtained across three clinics. Hearing aids representing commonly prescribed devices in these clinics were programmed using the final fitting parameter settings for clinical patients. Output from the hearing aids was recorded in an anechoic test box and/or using an acoustic manikin for sentences presented at three input levels (55 dB SPL, 65 dB SPL, 75 dB SPL), mixed with babble at four different SNRs (0 dB, +5 dB, +10 dB, Quiet). An acoustic metric based on signal envelope was used to quantify modification in the processed signal with respect to a quiet reference. Results to date indicate a wide distribution of signal modification across clinical hearing aid fittings, and show that there is greater signal modification at lower SNRs and higher input levels. The amount of signal modification is interpreted in the context of audiological factors (e.g., audiogram), as well as device-related factors (e.g., hearing aid coupling and manufacturer). Results provide a better understanding of the acoustic effects of clinically fit hearing aid signal processing. The translational nature of this work bridges the gap between discrete signal processing settings in the lab and real-world signal processing choices based on standard-of-care guidelines. [Work is funded by NIH.]

Friday, August 17

SESSION C5. Binaural Hearing Aids and Fitting

Session Chair: Anna Diedesch

5:00 PM  C5-O-1: Spatial hearing, hearing loss and hearing aids

* Virginia Best¹ and Jayaganesh Swaminathan²

¹ Boston University
² Starkey Hearing Research Center

Spatial hearing is important for environmental awareness, for segregating similar sounds, and for understanding speech in noise. It is often noted that hearing-impaired listeners have poor spatial sensitivity when compared to listeners with normal hearing, and it is assumed that this contributes to their speech communication difficulties. However, the data on this topic are inconclusive. When spatial deficits are reported in this population, it is often difficult to rule out conflating factors such as (monaural) audibility, or differences in age, as the root cause. Moreover, whether or not a deficit is observed seems to depend on the specific way in which spatial perception is measured. This talk will review some of our recent studies that have used speech stimuli to obtain more meaningful measures of spatial perception in listeners with normal and impaired hearing. These studies attempt to understand if and how spatial perception in real-world listening situations is affected by hearing loss, and what the consequences are for speech intelligibility. Clearer answers to these questions seem to be critical for fitting hearing aids binaurally and for informing future binaural processing strategies.
5:30 PM  C5-O-2: A semantic framework for binaural mechanisms in speech understanding

Benjamin Dieudonné and Tom Francart
KU Leuven – University of Leuven, ExpORL

It is well-known that hearing with two ears facilitates speech understanding in complex listening environments. Despite many years of research, binaural hearing remains to be a puzzling and intensely debated phenomenon. This discussion is often unintentionally semantic due to ambiguously defined binaural mechanisms, such as squelch and binaural interaction. Unfortunately, this may lead to poor interpretations of experimental results.

Therefore, we established a framework in which the following phenomena are explicitly defined and unambiguously related with each other: head shadow, redundancy, squelch, binaural interaction and spatial release from masking. We applied this framework to our own data and some data from existing literature.

The framework was able to facilitate the interpretation of experimental results. Most importantly, it naturally emphasized that a binaural benefit (better speech intelligibility with two ears than with one) is not necessarily due to spatial information: it might also be due to redundant (or complementary) information. Although this seems trivial, the use of spatial information to facilitate speech understanding is often being suggested in situations where it is probably absent.

We hope that the proposed framework will trigger researchers to explicitly define binaural phenomena when they are discussed, to avoid misunderstandings about the binaural system.

5:50 PM  C5-O-3: Effects of hearing loss and broad binaural fusion on dichotic concurrent vowel identification

Michelle Molis¹ and Lina Reiss
¹ VA RR&D NCRAR
² Oregon Health & Science University

Normally, listeners use frequency cues, such as fundamental frequency (voice pitch), to segregate sounds into discrete auditory streams. The ability to selectively attend to one talker or stream in the presence of other talkers or streams is crucial for speech perception in noise. However, many hearing-impaired (HI) individuals have abnormally broad binaural pitch fusion, such that tones differing in pitch by up to 3-4 octaves are fused between ears (Reiss et al., 2017). Broad binaural fusion leads to averaging of the monaural pitches (Oh and Reiss, 2017), and may similarly lead to fusion and averaging of speech streams across ears. In this study, we examined the relationship between binaural fusion range and speech fusion of dichotic speech stimuli, the simplest case of separating a “target” in one ear from a “masker” in the other ear. Specifically, dichotic vowel perception was measured in normally-hearing (NH) and HI listeners, with across-ear fundamental frequency differences varied. Synthetic vowels /i/, /u/, /a/, and /ae/ were generated with three fundamental frequencies (F₀) of 106.9, 151.2, and 201.8 Hz, and presented dichotically through headphones. For HI listeners, stimuli were shaped according to NAL-NL2 prescriptive targets. Although the dichotic vowels presented were always different across ears, listeners were not informed that there were any single vowel trials and were allowed to identify 1 vowel or 2 different vowels on each trial. When there was no F₀ difference between the ears, both NH and HI listeners were more likely to identify only 1 vowel. As the ΔF₀ increased, NH listeners increased the percentage of 2-vowel responses, but HI listeners were much less likely to do so—HI listeners were more likely to fuse two vowels together even with large ΔF₀. NH listeners with broad fusion also had poorer overall performance than those with sharp fusion. Confusion patterns were also compared with those obtained for concurrent vowel perception within a single ear. The patterns of confusions with dichotic vowels differ from those seen with concurrent monaural vowels, suggesting different mechanisms behind the errors. Together the findings suggest that broad fusion leads to spectral blending across ears, even...
for different $\Delta F_0$, and may hinder the stream segregation and understanding of speech in the presence of competing talkers. [Supported by NIH-NIDCD grant R01 DC013307 and the VA RR&D NCRAR C2361-C.]

6:10 PM C5-O-4: Can we trust our aided ears? - Acoustic directional information available in different hearing device styles

Florian Denk, Stephan Ewert, and Birger Kollmeier
Hearing4All & Universität Oldenburg

Despite the great benefit provided by modern hearing devices, poor spatial hearing abilities still remain a major problem. Even though the strongly reduced aided sound localization ability due to an unsuitable hearing device microphone placement has been described well in the literature, a quantitative analysis of the effect of different device styles and the individual pinna is still missing. This study presents a comprehensive database and a systematic objective analysis of these effects by assessing how well spectral directional cues are captured by the microphone(s) of different hearing devices in individual ears. We thus evaluate the directional information captured at 9 microphone locations that are integrated in 5 hearing device styles used both in a hearing aid and consumer device context. Data obtained in 16 human subjects and 3 dummy heads was evaluated. The underlying database of Head-Related Transfer Functions (HRTF) for the different hearing device microphones and device styles, as well as for the open ear, is publicly available (medi.uni-oldenburg.de/hearingdevicehrtfs/). Evaluation methods include previously established spectral distance metrics, a systematic assessment of qualitative aspects of the captured directional cues, and state-of-the-art models of human sound localization. Results confirm that full directional information is only captured with microphones located at the ear canal entrance or further inside the ear canal. Behind-the-ear microphones capture almost no spectral directional cues, and no considerable differences were noted between the 3 microphone positions of the utilized behind-the-ear device. When the microphone is placed in the cavum conchae in an in-the-ear type device, relevant spectral directional cues are captured. Nevertheless, errors against the open-ear HRTF that can be expected to deteriorate sound localization occur. The qualitative characteristics of these errors are discussed and compared to differences between HRTFs of individual humans’ ears. Furthermore, the spatial resolution of HRTFs of in-the-ear type hearing devices is poorer than those of the open ear. In consequence, even if perfect adaptation to the new directional cues available at the in-the-ear and behind-the-ear microphones is assumed, localization models predict a poorer localization performance than with the open-ear HRTFs. All results vary largely between individual ears, and results obtained with dummy heads do not perfectly reflect the data of human ears.

6:30 PM C5-O-5: Intelligibility, subjective preference and cortical measures for listeners using a binaural beamformer in realistic background noise at high signal to noise ratio

Jorge Mejia¹, Chloe Vella², and Isabella Moloney²
¹ The Hearing CRC / National Acoustic Laboratories
² Macquarie University

Listeners with hearing difficulties exert significantly greater effort than normal hearers when listening to talkers in loud background noise conditions. In such listening situations wearers of hearing aids benefit from directional microphones and, in high-end devices, binaural beamformers (BBF). On the other hand, in listening situations with low background noise levels, directional features are often switched off. However, there is no clinical evidence to suggest that switching off directional features is necessary or beneficial when the target speech is much greater than background noise level, i.e., high signal to noise ratio (SNR). The use of beamformers in high SNRs
was investigated for listeners attending a frontal target talker in realistic ambisonic noise. Here we present data from 2 preliminary examinations, in which two groups of listeners with normal hearing rated the sound quality of different microphone directivity schemes in realistic listening conditions. In the first study, an ‘idealised directivity’ scheme was created by manually adjusting the level of the ambisonic noise by 3 dB with omnidirectional microphones. The resulting enhancement of 3 dB SNR is equal to that commonly reported for cardioids relative to omnidirectional microphones in multi-talker speech conditions. The first group of 9 listeners were asked to perform two auditory tasks; a word recall test to ascertain speech understanding in noise performance and a subjective rating of preference and listening effort. At the same time $\alpha$, $\beta$ and $\gamma$ wave features from electroencephalogram (EEG) measures were recorded. These features were used as inputs to a neural network, which was trained to predict the speech-in-noise scores, rated preference and listening effort.

In the second study, the second group of 12 listeners rated the sound quality for omnidirectional, cardioid and a BBF, recently developed by the HEARing CRC. The 3 microphone types were tested in 4 realistic reverberant noise soundfields using ambisonics. For two of those noise fields, additional tests were conducted with noise levels individually adjusted for each participant to achieve 95% and 100% word recall. Subjective ratings were obtained for overall sound preference and ease of listening. Similarly, $\alpha$, $\beta$ and $\gamma$ wave features from EEG measures were also recorded. These features were then processed with the trained neural network from the first study, resulting in individual objective measures for each sound attribute tested. Outcomes are presented and discussed.

POSTER SESSION II
Paired with Sessions C3, C4, and C5
Friday 9:40 AM – 11:10 AM,
8:30 PM – 10:00 PM

Posters for Session II should be put up by 8:00 AM Friday, August 17, and taken down after 10:00 PM Friday, August 17. Presenters should be at their posters from 9:40 AM – 11:10 AM

C5-P-01: Spectral and binaural loudness summation: Essential for bilateral hearing aid fitting?
Monique Boymans1, Mirjam van Geleuken2, Maarten van Beurden1, Dirk Oetting2, and Wouter A. Dreschler2

1 Libra Rehabilitation and Audiology
2 Academic Medical Center
3 HörTech gGmbH

Aversiveness of loud sounds is a frequent complaint by hearing-aid users, especially when fitted bilaterally (Boymans et al. 2009, Hickson et al. 2010). Earlier research from Oetting et al (2016) indicated that the restoration of the narrowband loudness perception in hearing-impaired listeners may not be adequate for the perception of loud broadband signals, presented binaurally. This study investigates whether loudness summation can be held responsible for this finding.

The focus is on spectral loudness summation (loudness for broadband versus narrow-band signals), binaural loudness summation (loudness for binaurally versus monaurally presented signals) and binaural/spectral loudness summation.

In this study different aspects were investigated: (1) effect of different symmetrical hearing losses according to different classification of Bisgaard, (2) the effect of different spectral shapes of broadband signals. For the measurements we used “Adaptive Categorical Loudness Scaling” (ACALOS) (Brand and Hohmann, 2001) and loudness matching. Loudness matching was applied as a potentially faster technique to be used in a clinical setting.

Results show large individual differences for spectral and binaural loudness perception, especially for broadband stimuli. The large individual variability of
spectral and binaural loudness summation could not be predicted from the hearing loss configuration. The individual differences are that large that these should be considered during a hearing aid fitting procedure. The poster will discuss the option to use less time-consuming loudness matching procedures for the clinical applicability.

C5-P-02: Spatial release from masking in children with bilateral hearing loss: Effects of informational and energetic masking

Jenna Browning1, Emily Buss2, Mary Flaherty3, and Lori Leibold4
1 Boys Town National Research Hospital
2 The University of North Carolina at Chapel Hill

For listeners with normal hearing, masked speech recognition thresholds are lower when the target and masker are separated on the horizontal plane compared to when they are co-located. This effect is called spatial release from masking (SRM). While previous studies have shown that most children with hearing loss perform more poorly than age-matched children with normal hearing on measures of speech-in-speech recognition (Leibold et al. 2013), their ability to achieve SRM in the context of substantial informational and energetic masking has not been systematically investigated. In this experiment, SRM was measured for 13 children with hearing loss (ages 7-14 years), who were tested while wearing laboratory hearing aids. An additional 13 children with normal hearing served as controls (age-matched within ±6 months). An adaptive, open-set word recognition procedure was used to estimate 50% correct performance in the presence of a co-located or spatially separated masker. The masker was either speech-shaped noise or two streams of speech. The target speaker was located directly in front of the listener and the maskers were both presented at 0 degrees (co-located condition) or one masker was presented at +135 and the other at -135 degrees (spatially separated condition) on the horizontal plane. Speech recognition thresholds were lower for children with normal hearing than for children with hearing loss in both maskers and both conditions. Average SRM in the speech-shaped noise masker was 3 dB for children with normal hearing and 1 dB for children with hearing loss. Average SRM for the speech masker was 3 dB for children with normal hearing, but children with hearing loss did not obtain a significant SRM. Neither age nor audibility was associated with SRM in either masker for the children with hearing loss. The findings of this study indicate that children with hearing loss are at a disadvantage compared to their peers with normal hearing when listening to a talker in the presence of other talkers or noise sources occurring at different locations in the environment. Follow up studies are planned to evaluate the influence of hearing aid related factors, such as acclimatization and hearing aid signal processing, on SRM.

C5-P-03: Streaming of speech sounds based on localization cues under normal and impaired hearing

Marion David1, Olaf Strelicy2, and Andrew Oxenham3
1 University of Minnesota
2 Sonova A.G.

Hearing out one voice amid a complex auditory background is a real challenge for hearing-impaired (HI) listeners. Localization cues, such as interaural level and time differences (ILD and ITD) and spectral differences associated with the filtering operated by the head, pinna and torso, provide some of the cues needed for hearing out individual sources. The aim of this study was to investigate to what extent HI listeners can make use of these localization cues to segregate sequences of speech sounds into perceptual streams, in the absence of any simultaneous interference between the target and interfering sounds. The performance of HI listeners with and without their hearing aids were compared to that of older and younger NH listeners. Listeners were presented with sequences of speech sounds consisting of a fricative consonant and a voiced vowel (CV). The CV tokens were concatenated into interleaved sequences that alternated in positions in either the horizontal or median plane. The listeners were asked to attend only on sounds coming from in front of them, to ignore the other sounds, and to detect whether or not a repeated token was introduced within the attended sequence. In this way, performance was expected to improve as the listeners were better able to perceptually segregate the two sequences. The results suggest that 1) discrimination increased as the spatial separation increased in both horizontal and median planes; 2) age does not affect the ability to discriminate and segregate speech sounds based on spatial location; 3) HI listeners are able to make some use of localization cues to segregate speech sounds, even when wearing hearing aids; but 4) the front-back confusions observed in all groups may be exacerbated by the use of hearing aids.
In noisy and reverberant environments, hearing aids comfortably amplify sound and improve signal-to-noise ratios. This is accomplished by using direction microphones and complex signal processing such as wide dynamic-range compression (WDRC) and noise reduction. Sound localization cues, however, may be reduced, distorted, or potentially additive due advanced signal processing which could potentially interfere with communication in complex acoustic scenes. Specifically, WDRC and strong directional microphones are known to reduce interaural level differences (ILD). Whether interaural time differences (ITD) are distorted by amplification is less clear. We had speculated that ITD would be distorted due to minute signal processing delays or multi-path acoustics. Our previous results, however, showed no difference in ITD for receiver-in-the-canal hearing aids, compared to unaided, with the exception of when bilateral beamforming was activated (Diedesch et al. 2017, JASA, 141(5):3638). Here, we evaluated premium hearing aids from major manufacturers to verify reduced ILD and unaltered ITD, compared to unaided recordings, across different signal processing algorithms. To further investigate spatial cue distortion, recordings were evaluated in two time-windows, “onset” and “ongoing,” and compared to full-wave analysis (Diedesch et al. 2017). Binaural acoustic recordings were collected on an acoustic manikin fit with spondees. Recordings from anechoic and simulated rooms were analyzed for frequency-specific ITD and ILD. This work was supported by the Acoustical Society of America’s F.V. Hunt Postdoctoral Research Fellowship and Western Washington University.

References:
C5-P-06: Subjective evaluation of binaural noise reduction and cue preservation algorithms in a cocktail party scenario
Daniel Marquardt\textsuperscript{1}, Ivo Merks\textsuperscript{1}, Tao Zhang\textsuperscript{1}, and Simon Doclo\textsuperscript{2}
\textsuperscript{1} Starkey Hearing Technologies
\textsuperscript{2} University of Oldenburg

Due to their decreased ability to understand speech in the presence of background noise and/or interfering speakers, hearing impaired listeners may encounter severe difficulties communicating with other people in a cocktail party scenario. To improve speech understanding for hearing aid users, many different noise reduction algorithms have been developed over the years. Nowadays, binaural wireless hearing aids also allow these algorithms to exploit the microphone signals of both hearing aids. While some state-of-the-art binaural algorithms, e.g., the binaural minimum variance distortionless response (BMVDR) beamformer, perform optimally in terms of noise reduction and binaural cue preservation of the target speaker, they distort the binaural cues of the interfering speaker(s). This causes the hearing aid user to perceive the target speaker and the interfering speaker(s) as coming from the same direction, prohibiting the auditory system of the hearing aid user to use its own binaural squelch mechanism. Aiming at preserving the binaural cues of the interfering speaker(s), several binaural algorithms, e.g., the binaural linearly constrained minimum variance (BLCMV) beamformer [1], have been recently developed. In comparison to the BMVDR beamformer, these algorithms however result in a lower noise reduction performance.

In this contribution, we subjectively compare the performance of two versions of the BMVDR and the BLCMV beamformer, either maximizing the signal-to-noise-ratio (SNR) or the signal-to-interference-and-noise-ratio (SINR), in terms of speech intelligibility improvement and subjective preference. For a reverberant acoustic scenario (T60 = 300 msec) comprising one target speaker, one interfering speaker and diffuse background noise, a listening test was performed with N=14 normal-hearing subjects. On the one hand, when maximizing SNR, the BLCMV beamformer, which preserves the binaural cues of the interfering speaker, is able to significantly improve speech intelligibility compared to the BMVDR beamformer. Similarly, when maximizing SNR, the results of the preference test show that the BLCMV beamformer is generally preferred over the BMVDR beamformer, depending on the input SINR and the spatial position of the speakers. On the other hand, when maximizing SINR, the BLCMV neither improves speech intelligibility nor subjective preference compared to the BMVDR. These results show that the performance of binaural algorithms largely depends on the acoustical scenario and the optimality criterion.

Reference:

C5-P-07: Restoring normal-hearing performance in noise for hearing-impaired listeners using binaural beamformer
Richard van Hoesel\textsuperscript{1}, Anna O’Brien\textsuperscript{1}, and Jorge Mejia\textsuperscript{1,2}
\textsuperscript{1} The Hearing CRC
\textsuperscript{2} National Acoustic Laboratories

A new binaural beamformer developed at the Hearing CRC, Australia, was evaluated in moderately hearing-impaired listeners attending target speech in spatially distributed multi-talker noise. The beamformer is specifically designed to provide robust SNR improvements and be tolerant of modest target misalignment, without introducing objectionable artefacts or modifying spatial impression for either targets or surrounding noise. Computational complexity is very low to enable implementation on a wide range of hearing devices.

The beamformer was compared to omnidirectional and traditional cardioid directional microphone schemes. Assessment included speech reception thresholds for 50% correct intelligibility (SRT50). In addition to targets being presented from directly in front of the listener, SRT50 was also measured for a deliberate target misalignment of 10°. Access to spatial hearing cues was evaluated for the cardioid and beamformer schemes using a sound-source direction identification task in noise, which employed nine loudspeakers spanning the frontal 180-degree hemifield. Acceptable noise levels (ANLs) and subjective ratings in noise were measured using fixed-level targets to the front for all three directional schemes. Additional subjective ratings were determined for a two-talker dialogue in noise, with the two talkers presented either from the same location, or separated by 45° or 67°. The two separated conditions address the concern that beamformers may be undesirable in situations where target location is not fixed to the front of the listener. Comparative SRT-50 and ANL measures were also obtained for unaided listeners with normal hearing. Results show that when using the beamformer, the hearing-impaired listeners performed at least as well as the unaided normal-hearing
listeners on tests of speech intelligibility and tolerance to noise. For the hearing-impaired listeners, the beamformer provided robust benefits over traditional cardioid directivity, even when head orientation was purposefully misaligned by about 10 degrees from the target direction, and did so without degrading sound localization. Subjective preference for the beamformer over the cardioid was retained when listeners were free to turn their head whilst attending a conversation (in noise) between two talkers separated by 45°. At a larger talker separation of 67°, the beamformer was still rated at least as high as the cardioid. The subjective-rating and ANL benefits for the beamformer over the cardioid were comparable to the same benefits for the cardioid over the omnidirectional directivity. Combined these outcomes clearly indicate that hearing-impaired listeners in noisy environments are likely to benefit from the use of the CRC binaural beamformers.

C5-P-08: Analysis of speech intelligibility and preference for binaural beamforming
Ivo Merks, Jumana Harianawal1, and Kyle Walsh
Starkey Hearing Technologies

Improving the signal to noise ratio (SNR) for listeners with hearing impairment has long been a goal of various amplification schemes. One of the most common ways to improve listening in background noise is by using first order microphones or microphone arrays within one hearing aid (called bilateral hearing aids) or with the advent of wireless technology, microphones or microphone arrays within two hearing aids (called binaural beamforming hearing aids).

Although the theoretical SNR improvement of binaural beamforming is 2-3 dB, the improvement in speech intelligibility is lower because of subject-dependent in-situ SNR, acoustic leakage (i.e. venting), and the distortion of binaural cues. Furthermore, subject preference for binaural solutions might be limited due to additional latency, wireless processing, and the limited access to binaural cues. This paper will analyze how these different factors impact speech intelligibility as well as preference for binaural beamforming.

Electro-acoustic evaluation: The in-situ SNR improvement has been measured on 8 subjects as well as KEMAR with hearing aids in 4 modes: mute, omni, directional, and binaural beamforming. The results show that on average, the SNR improvement of the binaural processing is 0.5 dB lower on subjects than on KEMAR. Furthermore, the acoustic leakage lowers the SNR improvement by 0.5 dB.

Speech recognition test: The goal of this part of the study was to estimate the effect size of three different factors on the speech recognition threshold (SRT). The factors were binaural processing, wireless processing, and leakage/latency. The SRT of 15 normal hearing subjects was measured using the hearing in noise test (HINT) and the results showed that binaural processing improves SRT by 1.03 dB (significant) acoustic leakage worsens SRT by 0.91 dB (significant), while wireless processing and latency did not have a significant effect.

Preference test: The goal of this part of the study was to estimate the effect size of different factors (binaural processing, wireless processing, and latency) on the subjects’ preference. 11 normal hearing subjects and 16 hearing impaired individuals indicated preference for the three factors for external stimulus and own voice in the presence of background noise. The results showed that the subjects had preference for the binaural processing but no preference for the wireless processing or latency.

C5-P-09: A Wearable Platform for Hearing Aids Research
Louis Pisha1, Sean Hamilton2, Dhiman Sengupta2, Ching-Hua Lee3, Krishna Chaithanya Vastare3, Sergio Luna4, Tamara Zabatty5, Cagri Yalcin6, Alex Grant5, Mark Stambaugh4, Arthur Boothroyd4, Ganz Chockalingam3, Rajesh Gupta7, Bhaskar Rao1, and Harinath Garudadri5
1 Dept. of Electrical and Computer Engineering, University of California, San Diego
2 Dept. of Computer Science and Engineering, University of California, San Diego
3 Dept. of Mathematics, University of California, San Diego
4 Dept. of Cognitive Science, University of California, San Diego
5 Qualcomm Institute, University of California, San Diego
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Overview: In this contribution, we describe the wearable version of the Open Speech Platform (OSP), based on technology from modern smartphones. We present the system architecture and discuss salient design aspects in support of such investigations. OSP has rapid prototyping capabilities for audiological and speech sciences investigations in the lab and in the field. The platform also facilitates the development of new hardware, software, and signal processing algorithms for hearing aids (HAs), hearing assisted devices, hearables, and other realtime speech or audio
We conducted two experiments on speech perception by hearing-impaired (HI) listeners aided with binaurally-linked dynamic range compression (DRC) hearing aids. The linked DRC preserved interaural level difference (ILD) cues for spatially separated sources. In a first experiment, word identification was tested using coordinate response measure (CRM) sentences presented sequentially from five loudspeakers in a sound-treated room. Testing was conducted with and without a simulated single reflection, because acoustical analysis demonstrated distortion of reflections when using independent DRC at the two ears. Word identification scores of 13 HI participants showed neither an effect of DRC linking nor of the simulated reflection. In a second experiment with 14 HI participants, the CRM sentences were high-pass filtered at 1 kHz to reduce cue redundancy, and sentences were presented sequentially and simultaneously. In these high-pass filtered conditions, linked DRC resulted in significantly higher identification scores than independent DRC by 12% and 6% in sequential and simultaneous presentation, respectively. Overall, these results suggest that objective benefits of linked DRC are most apparent for speech materials and listening situations where high-frequency information is needed for identification.

C5-P-11: On the potential benefits of audibility in the 5-10 kHz region for orientation behaviour

William M Whitmer\textsuperscript{1}, Suzanne C Levy\textsuperscript{2}, David McShefferty\textsuperscript{1}, W. Owen Brimijoin\textsuperscript{1}, and Graham Naylor\textsuperscript{3}

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\textsuperscript{3} University of Nottingham

Current hearing aids have a limited bandwidth, which limits the intelligibility and quality of the devices, and inhibits their uptake amongst the millions who could benefit from them. Recent advances in signal processing as well as novel methods of transduction (e.g., fibre-optic stimulation of the tympanic membrane), however, allow for a greater useable frequency range. Previous studies have shown a benefit for this extended bandwidth in consonant recognition, gender identification and separating sound sources. The question remains on whether there would be any direct spatial benefits to extending bandwidth. To explore this question, we used a dynamic localization method to capture myriad orientation benefits in a realistic situation.

To create a plausible scenario, we used a near-far distinction between targets and distractors speaking at

C5-P-10: Effects of binaurally-linked dynamic range compression on word identification by hearing-impaired listeners

Olaf Strelcyk\textsuperscript{1}, Pavel Zahorik\textsuperscript{2}, Chhayakant Patro\textsuperscript{3}, and Peter Derleth\textsuperscript{4}

\textsuperscript{1} Sonova US
\textsuperscript{2} University of Louisville
\textsuperscript{3} Heuser Hearing Institute
\textsuperscript{4} Sonova AG

We are interested in enabling both conventional audiological investigations and new studies involving innovative algorithms or clinical elements. Thus, there is a focus on replicating the functionality of existing hearing diagnosis, treatment and assessment systems on a portable, open platform, as well as providing extensibility and features far beyond the capabilities of current systems to enable new discoveries in hearing healthcare.

**Hardware:** The wearable unit runs OSP Release 2018a on top of 64-bit Debian Linux for binaural HA with an overall latency of 5.08 ms using only 21% of the CPU resources. The ear-level assemblies are of the behind the ear, receiver in the canal (BTE-RIC) form factor and connect to the sound processing unit via cables. The ear-level assemblies support up to 4 microphones on each ear: front, rear, in-ear, and Voice Pick-Up (bone conduction). The four microphones and receiver are connected to an Analog Devices ADAU1372 codec, which is a high-quality but inexpensive consumer audio codec that includes microphone preamplifiers and headphone drivers, and which samples all channels at 96kHz 24bit. The wearable unit (which also includes the battery) hosts a Qualcomm Snapdragon 410c chipset with a quad-core 64-bit ARM CPU, GPU, DSP, and a variety of peripherals.

**Software:** The HA sound processing software includes 6 channel processing, wide dynamic range compression, adaptive feedback cancellation and speech enhancement as described in Release 2018a. In addition, the wearable unit hosts an embedded web server (EWS) with abilities to monitor and control the HA state in realtime. The EWS also provides application programmer interfaces (APIs) to access speech and noise statistics, GPS, user state through ecological momentary assessments (EMA), etc. in HTML and PHP. We demonstrate typical audiological studies implemented as web apps and administered by browser enabled devices.
equivalent levels in a simulated large room with common building materials. Twenty-eight mostly normal-hearing adult participants reoriented themselves as quickly and accurately as comfortable to a new nearfield talker continuing a story in a background of farfield talkers of the same overall level. All stimuli were either low-pass filtered at 5 or 10 kHz on each trial to simulate current or novel technologies, respectively. To further simulate current hearing aids, participants wore microphones above the pinnae and insert earphones adjusted to provide a linear, zero-gain response.

Previous assessments of spatial benefit rely solely on the accuracy of locating a sound source. Here, infrared motion-tracking allowed each individual trajectory to also be analysed for velocity, reversals, misorientations, complexity, start and end time. Results across listeners showed a significant increase in velocity and significant decrease in start time (both $p < 0.0001$). These earlier, swifter orientations demonstrate spatial benefits beyond accuracy in plausible conditions; the extended bandwidth afforded by new hearing technologies provided more salient cues in a realistic mixture of talkers. These results not only bolster the efficacy of such technology, but also help establish new measures of benefit on which future technology can be developed and evaluated. [Work supported by EarLens, the Medical Research Council (grant number U135097131) and the Chief Scientist Office of the Scottish Government.]

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**Saturday, August 18**

**SESSION D6, Signal Processing for Hearing Aids**

Session Chair: Chas Pavlovic

**8:00 AM  D6-O-1: Combining remote microphones with hearing aid processing**

*James Kates and Kathryn Arehart*
*University of Colorado*

This study investigates the interaction of remote microphones with hearing aids. In a remote microphone (RM) system, the talker is provided with a microphone that transmits a signal directly to the listener’s hearing aids, thus bypassing the room noise and reverberation. The RM signal, however, also bypasses the acoustics of the head and pinna, creating an apparent sound source that is lateralized at the center of the head rather than localized at the talker. Externalization of the RM signal is greatly improved by simulating the acoustic localization cues in combination with early room reflections, and previous work has documented this benefit for headphone presentation using linear amplification to compensate for the hearing loss. However, integrating the RM with a hearing aid presents a more complicated scenario. The RM signal is first combined with the acoustic inputs at the hearing-aid microphones, and this combined input is processed through hearing-aid algorithms such as wide dynamic-range compression. In addition, the hearing-aid output is modified by the earmold acoustics and mixed with the acoustic signal that is transmitted through the vent or open fitting. This study reports listener data on externalization, speech intelligibility, and speech quality for RM signals that have been modified to improve externalization when the signals are presented through a realistic binaural hearing-aid simulation. Parameters varied in the experiments include the blend of the RM and hearing-aid microphone signals, type of hearing-aid processing, and venting. [Research funded by a grant to the University of Colorado from GN ReSound.]
**D6-O-2: A deep learning-based segregation algorithm to improve speech intelligibility of hearing-impaired listeners in reverberant-noisy conditions**

*Yan Zhao, DeLiang Wang, Eric Johnson, and Eric Healy*
*Ohio State University*

Recently, deep learning-based speech segregation has been shown to improve human speech intelligibility in noisy environments. However, one important factor not considered is room reverberation, which characterizes typical daily environments. Reverberation and background noise have confounding effects and severely degrade speech intelligibility of hearing-impaired (HI) listeners. In this study, a deep learning-based time-frequency masking algorithm is proposed to address both room reverberation and background noise. Specifically, a deep neural network (DNN) was trained to estimate the ideal ratio mask (IRM), where anechoic-clean speech was considered as the desired signal. Then, the estimated ratio mask was applied to the magnitude spectrum of reverberant-noisy speech to obtain enhanced speech.

Intelligibility testing was conducted under reverberant-noisy conditions with reverberation time T60 = 0.6 s, plus speech-shaped noise (SSN) and babble noise at various signal-to-noise ratios (SNRs). The experiments demonstrated that substantial speech intelligibility improvements were obtained for HI listeners. Specifically, group mean algorithm benefit for the HI subjects ranged from 22 to 33 percentage points in SSN (5 to -5 dB SNR), and from 19 to 47 points in babble (10 to 0 dB SNR). For the normal-hearing (NH) subjects, group mean benefit was smaller, but reached 13 percentage points in 0 dB SNR babble. It is worth noting that sentence intelligibility scores of HI listeners with algorithm processing approached those of young-adult NH listeners without processing.

From the perspective of hearing aids, the monaural nature of the algorithm provides inherent convenience in device implementation compared to microphone-array techniques. The supervised learning framework shifts much of the workload to the training stage, and during the operational (test) stage, the algorithm involves only feature extraction and frame-level ratio masking using a trained DNN, both of which can be performed efficiently and are amenable to real-time implementation.

To our knowledge, these test results provide the first demonstration of speech intelligibility improvements by a monaural algorithm for reverberant-noisy speech. As reverberation and background noise both occur in typical listening situations, this study represents a major step towards deploying deep learning algorithms to help the speech understanding of HI listeners in everyday environments [Work supported by NIH].

**D6-O-3: Deep neural networks for noise reduction under hearing aid side conditions**

*Marc Aubreville¹,², Kai Ehrensperger², Tobias Rosenkranz¹, Benjamin Graf⁰, Henning Puder¹, and Andreas Maier²*

¹ Sivantos GmbH
² Friedrich-Alexander-Universität Erlangen-Nürnberg

Noise reduction has been added to the feature set of commercially available hearing aids a long time ago. However, they are typically only able to handle special noise, such as stationarity noise, strong impulsive noise or noise originating from certain directions, e.g. the back. The performance of these current algorithms is strongly limited by two main factors: the feasibility to model a complex background scenario and technical realization constraints. While the constraints for computational complexity are being reduced with improving battery technologies and smaller structural sizes in chip manufacturing, the diversity of background noises poses significant challenges towards detection and reduction algorithms.
In automatic speech recognition systems, where power constraints play a less significant role, deep learning-based schemes have been introduced and are applied in productive use. The big advantage of these schemes is that no explicit model knowledge of all potential background noises or target signals is required, but instead a large data set is used for training, allowing the model to learn the signal enhancement function in dependence of noisy input signals on its own. These schemes, on the other hand, have long been criticized as being black box approaches with unpredictable behavior in unknown situations. This is especially true for a direct prediction of the enhanced signal which could produce unpleasant or annoying signals at the ear drum. To cope with the risk of potential misbehaviors, one of the possibilities is to embed a noise reduction operator into an existing signal processing framework.

In this work, we present and evaluate a deep neural network-based noise reduction scheme that can be integrated seamlessly into existing hearing instrument signal processing chains. Our approach complies with hard, physical hearing aid constraints such as strictly limited latency and operates on signals from a state-of-the-art hearing instrument filter bank. By predicting a noise reduction gain vector, it can be combined with traditional noise reduction schemes. Besides an objective evaluation and results of a subjective test, we will present examples from our test data set (mixed at defined signal to noise conditions) as well as from completely unrelated real-world audio mixtures.

9:00 AM  
D6-O-4: Accessible infrastructure for hearing research: A commodity-hardware-based mobile prototype of a hearing aid featuring the openMHA.org research software platform

Marc-Rene Schaedler1, Florian Denk1, and Birger Kollmeier1,2
1 Universität Oldenburg
2 Hearing4All

Scientific progress, the validation of experimental results, and the adoption of new solutions depend on the availability of knowledge and tools. However, no access to the internals of hearing aids is available for users, students, and researchers. This is mainly due to limitations of the hardware, due to liability issues of medical devices, and due to economic interests (protected IP). Many of these limitations are not critical for research purposes. However, currently no common accessible platform has been established for hearing aid research (in contrast to other “digital hearing” disciplines, such as, e.g., automatic speech recognition). The recent publication of a free and open research platform for hearing aid algorithms, the open Master Hearing Aid (MHA), potentially enables mobile research devices to which the restrictions mentioned above do not apply. The openMHA platform bundles over 15 years of development of software tools for hearing research. A prototype of a wearable hearing aid is introduced here which is based on commodity hardware (including a Raspberry Pi 3 SOC, a low-latency stereo sound card, and in-ear headphones with integrated binaural microphones). It runs the openMHA in real-time. The whole setup fits in a belt bag, weighs less than 500g, operates several hours without charging, and is configurable via WIFI. Latencies of less than 10ms could be achieved running openMHA on Raspbian Linux with the stock Linux kernel. Instructions for building the prototype as well as an SD-card image with a pre-configured software environment is provided on a website (https://github.com/m-rs/hearingaid-prototype/) which aims to serve as a community hub for the hardware project. Further progress and research results can be shared and easily incorporated in updated versions. The prototype may serve as a baseline for comparisons with real devices in field test, as a reference for new software (e.g., noise reduction) or hardware (e.g., microphone) developments, or it may be employed for teaching purposes. Many of the fundamental problems with hearing devices can be studied in the field including latency, calibration, fitting, effect of occlusion, acoustic feedback, audible artifacts, speech and music perception. The affordable and flexible hardware allows establishing an infrastructure that enables the rapid implementation and unrestricted distribution of new algorithms for testing under field conditions across labs and beyond the usual target groups. It has the potential to substantially
grow the community of hearing technology developers and testers, and hence, to increase the pace of development.

POSTER SESSION III
Paired with Sessions D6, D7, and D8
Saturday 9:30 AM – 11:00 AM, 8:30 PM – 10:00 PM

Posters for Session III should be put up by 8:00 AM Saturday, August 18, and taken down after 10:00 PM Saturday, August 18. Presenters should be at their posters from 9:30 AM – 11:00 AM

D6-P-01: Influence of signal enhancement algorithms on auditory movement detection in acoustically complex situations
Micha Lundbeck¹, Giso Grimm¹, Volker Hohmann¹, Lars Bramslow², and Tobias Neher³
¹ University of Oldenburg
² Eriksholm Research Centre
³ University of Southern Denmark

So far, little is known about how hearing loss and hearing aid (HA) signal processing impact spatial hearing and/or movement perception in complex environments. Previously, we showed that concurrent distractor sounds impair the detectability of left-right source movements and reverberation that of near-far source movements for older hearing-impaired (OHI) listeners (Lundbeck et al., 2017). In a headphone-based follow-up study, we investigated ways of improving these deficits with computer-simulated HA processing (Lundbeck et al., 2018a). To that end, we examined the impact of two beamforming algorithms and a binaural coherence-based noise reduction scheme on the acoustic cues underlying movement perception. We found that the applied processing led to greater monaural spectral changes as well as increases in signal-to-noise ratio and direct-to-reverberant energy ratio in our test stimuli. Furthermore, we found that it partly restored source movement detection for OHI listeners in the presence of distractor sounds. In a third study, we extended these findings towards equivalent measurements made with a loudspeaker array and wearable HAs (Lundbeck et al., 2018b). For a group of 13 OHI listeners, we found no differences in left-right or near-far target movement detectability. In this contribution, we discuss the results from this project with a focus on the potential and difficulties associated with measuring spatial awareness perception in complex acoustic environments.

References:

D6-P-02: Head shadow enhancement to improve sound localization and speech intelligibility in noise
Benjamin Dieudonné and Tom Francart
KU Leuven – University of Leuven

Head shadow attenuates sounds from the contralateral side for each ear. On the one hand, this introduces interaural level differences (ILDs) which can be used to localize sounds in the horizontal plane. On the other hand, it can attenuate noise sources such that it improves speech understanding. Unfortunately, (1) this behaviour is relatively weak for low frequencies, and
(2) ILDs vary non-monotonically as a function of the angle of incidence of sound, such that they cannot be used to localize sounds unambiguously. Therefore, (1) listeners with high-frequency hearing loss experience a limited benefit from head shadow, and (2) ILDs cannot fully substitute for interaural time differences (ITDs) to localize sounds for listeners with poor ITD sensitivity. Typical bimodal listeners (cochlear implant listeners with a contralateral hearing aid) lack both high frequency hearing (in the non-implanted ear) and temporal fine structure sensitivity (in the implanted ear), such that they are among the worst performers in localization and speech intelligibility in noise.

We developed a new method to artificially enhance the head shadow effect in low frequencies, i.e., to increase contralateral attenuation and to make ILDs monotonic as a function of angle of incidence. Head shadow enhancement is achieved with a fixed beamformer with contralateral attenuation in each ear. Each beamformer uses a microphone array consisting of two microphones, achieved with one microphone per device and a link between both devices.

In an experiment with simulated bimodal listeners and pre-processed stimuli, we found that head shadow enhancement significantly improved localization performance and speech intelligibility in spatial conditions where head shadow is expected to be present. Next, we developed a real-time implementation to validate the method on actual bimodal listeners. Results of the localization and speech-in-noise intelligibility experiments with this setup will be presented at the conference. Apart from bimodal listeners, the method is also promising for bilateral cochlear implant and hearing aid users. Its computationally cheap processing makes the method suitable for application in current clinical devices.

D6-P-03: Comparison of binaural MVDR-based beamforming algorithms using an external microphone

Nico Gößling and Simon Doclo
University of Oldenburg

Besides reducing undesired sound sources, an important objective of a binaural noise reduction algorithm is to preserve the spatial impression of the acoustic scene for the hearing aid user, such that no mismatch between acoustic and visual information occurs and the binaural hearing advantage can be exploited. Although the binaural minimum variance distortionless response (MVDR) beamformer is able to preserve the binaural cues of the target speaker, it distorts the binaural cues of the background noise. Hence, several extensions have been proposed, aiming at preserving the binaural cues of the background noise [1]. Because the performance of noise reduction algorithms is partly limited by the physical design, requiring the microphones to be integrated into the hearing devices, the usage of an external microphone that is spatially separated from the head-mounted microphones has been recently explored [2-4]. It has been shown that an external microphone enables to improve both the noise reduction performance as well as the binaural cue preservation performance.

In this contribution, we present a comparison of several binaural MVDR-based beamforming approaches that make use of an external microphone. First, the external microphone is merely used to provide an estimate of acoustic variables (e.g., relative transfer functions) that are required for the binaural beamforming algorithms. Second, the external microphone is used in conjunction with the head-mounted microphones as an additional input signal. Using realistic recordings of a moving target speaker in a reverberant room the performance of the considered binaural beamforming approaches is compared in terms of noise reduction performance (using speech-intelligibility-weighted SNR) and binaural cue preservation (using reliable ITD and ILD cues from an auditory model).

References:
In recent decades, advances in hearing aid technology have been focused on the processing of the information received by the microphones—hence ‘digital’ hearing aids—without regard to the microphone’s relatively dated and static designs. The basic hearing aid system has always been a fixed feed-forward system, with the acoustic signal passing from microphone to amplifier (digital signal processor) to receiver (loudspeaker). The microphone picks up the signal, the processor digitises the sound, enhances it for the impaired system (e.g. by compressing and amplifying the signal into the wearer’s limited dynamic range), and converts it through the receiver into a form of intelligible signal to the human auditory system. Nevertheless, this is a process mainly relied on power-demanding digital machinery requiring signal conversions (analogue-to-digital and vice-versa), buffering and computing arithmetic operations imposing time delays within the signal chain. There is, therefore, huge potential for new approaches to hearing prostheses based on novel microphones and signal processing frameworks. In this research work, we consider if microphones could be designed to be sensitive only at selected frequencies of interest, whilst also providing frequency agility with adaptive sensitivity in order to track specific signals of interest. The aim is to present a novel concept of signal processing performed at the sensor level. A real-time acoustic signal processing framework integrating a frequency selective MEMS microphone placed in a closed-loop system was engineered to support the concept of a fully-adaptive sensory-system. This is a concept where the transducer becomes part of the signal processing chain by exploiting feedback control mechanisms (DC and AC) between mechanical (microphone’s diaphragm) and electrical (analogue front-end) systems that together can enhance peripheral signal processing, while changing the effective electrical mechanical characteristics of the MEMS microphone. The goal is to improve the performance of the signal processing framework within a hearing aid system by: (1) reducing the number of DSP operations used within a sound processor so that the power consumption of those computational units might be less demanding; (2) decreasing the signal processing latency by reducing both the amount of data processing and exchange between digital buffers, eliminating numerous issues due to the delay and level limiting (distortion) incurred from signal conversions and computation (round-off errors), thus increasing the fidelity of the signal; (3) moving the DSP operations to one side to allow the use of more sophisticated signal-detection and noise-reduction algorithms.

One of the most significant challenges for hearing-impaired (HI) listeners is understanding speech in noisy environments. Unfortunately, current hearing aids are often ineffective in these situations. Single-channel speech enhancement algorithms such as Wiener filtering or the spectral subtraction technique have been widely used for hearing aids to deal with noisy speech. Many single-channel techniques can improve the comfort of listening in noisy environments. However, speech intelligibility for HI listeners is not always improved by these classical speech enhancement techniques. Recently, machine-learning-based single-channel speech enhancement algorithms have been reported to improve the intelligibility of noisy speech for HI listeners. For instance, Wang and co-workers proposed a deep neural network-based speech enhancement (DNN-SE) algorithm with time-frequency masking [1,2]. The DNN-SE algorithm showed high performance in terms of short-time objective intelligibility (STOI) and perceptual evaluation of speech quality (PESQ). They also reported that improved intelligibility was observed for both HI listeners and normal-hearing (NH) listeners even in negative-SNR environments. In this work, we fabricated a prototype of the DNN-based single-channel real-time speech enhancement system to discuss the performance in a real environment and problems to be overcome for practical use. The algorithm was implemented on a laptop computer (DELL: Inspiron) by C++ language with real audio I/O. The DNN was trained to learn the log-power spectra of clean speech for a given spectra of noisy speech, and was constructed with two hidden layers (with each layer containing 513-380-257 neurons). Ten hours of the JNAS speech dataset, which is a speech corpus of newspaper article sentences read in Japanese, was used to train the DNN. The total latency of the system was about 42 ms (algorithmic delay: 32 ms; hardware-dependent delay: about 10 ms). Finally, a subjective
evaluation experiment on NH listeners was carried out using the prototype.

References:

D6-P-06: An algorithm for reverberation suppression in cochlear implants
Kostas Kokkinakis and Josh Stohl MED-EL

While a high level of speech understanding in quiet is attainable by many cochlear implant (CI) recipients, speech perception in reverberant environments remains a challenge for most CI listeners. In fact, intelligibility of reverberant speech declines exponentially with a linear increase in reverberation time (RT60), even in quiet environments. Adding reverberant energy to a speech signal in either quiet settings or in environments containing noise disrupts temporal envelope variations and produces distortions to the fine structure information, thus hindering speech perception. In this study, a front-end signal processing algorithm for reverberation suppression was tested with MED-EL CI users in simulated noisy environments. The reverberation suppression strategy uses a selection criterion to adjust a tunable mask based on an online estimate of the inherent coherence statistics calculated between two microphones placed on either side of the head (inter-microphone coherence). The estimated time-frequency mask is applied to the spectro-temporal bins of the corrupted acoustic input to remove the additive reverberant energy present in the signals recorded from the two microphones. The mask estimation stage does not require access to an ideal clean or an uncorrupted signal, and therefore this solution is ‘blind’ and generalizable to any acoustic environment. To evaluate the overall clinical potential of this strategy, a total of ten post-lingually deafened adult CI recipients implanted with MED-EL devices participated in this study. All subjects were tested unilaterally with their clinical processor settings, with and without the additional front-end dereverberation algorithm. Speech reception thresholds (SRTs) were measured in four-talker babble, which is a perceptually complex stimulus that can produce both energetic and informational masking. Pre-recorded head-related impulse responses (HRIRs) were used to simulate two different realistic listening conditions with reverberation times equal to RT60 = 0.05 s and 0.6 s. Using the HRIRs, the target speech was presented from the front and the single competing four-talker babble was presented ipsilateral to the implanted ear. SRTs were measured with both HRIRs with the dereverberation strategy either enabled or disabled, yielding a total of four different experimental conditions. The speech stimuli were presented via direct stimulation using the RIB2 research library and the MAX hardware interface. The benefit observed due to the dereverberation algorithm was calculated by subtracting the SRTs obtained in the strategy-enabled conditions from the SRTs observed in the strategy-disabled configurations. A consistently large benefit was observed due to the front-end processing algorithm across all conditions tested.

D6-P-07: Modulation spectrum as an additional quantifier of hearing aid processing
Petri Korhonen Widex ORCA

While an acoustic signal is quantified by its three dimensions - frequency, intensity and time, much of the attention in hearing aid dispensing has been invested in the optimization of the output in the frequency and intensity dimensions. This is evidenced by the fact that the standards issued by the ANSI (S 3.22, ANSI, 2003) on hearing aids have stringent requirements on measuring/reporting the frequency-intensity output of the hearing aids. The only measurement of temporal processing is the measurement of attack and release time using a sinusoidal signal. Advanced hearing aid features such as wide dynamic range compression (WDRC), digital noise reduction (DNR), and directional microphone attempt to improve audibility or enhance signal-to-noise ratio (SNR). Unfortunately, the deliberate alteration of the signal to optimize audibility and/or SNR can lead to inadvertent alteration of the temporal characteristics of the input signal. The alterations on the temporal envelope of the signal may create suboptimal listening conditions, particularly to listeners with greater degrees of hearing loss or reduced working memory capacity. In the absence of a formalized tool to quantify temporal envelope changes, a measurement of modulation index derived from modulation spectrum of the input can be used to study the changes in modulation characteristics as a consequence of the signal processing in hearing aids. Measurement of modulation spectrum allows the use of natural speech signal in quiet or in noise at levels, which activate the advanced hearing aid features in a
manner that is expected in real life use. In this study we will report data from the measurements that evaluated the effects of individual hearing aid features on modulation index, and will also show how modulation index may be used to compare effects of hearing aid processing on modulation characteristics between commercial hearing aids. Effects of hearing aid processing were measured with continuous ISTS speech signal presented at 60, 70, and 80 dB SPL in quiet and in the presence of speech shaped background noise presented from the front or the back. The results demonstrated that modulation spectrum is effective at investigating combined effects of processing on temporal envelope. The measure showed differences in temporal processing in commercial devices despite little differences observed using standard test-box measures. The measurement of modulation spectrum can therefore provide additional insight on the effects of hearing aid processing on temporal envelope that traditional test-box measures may not capture.

D6-P-08: Effects of hearing-aid amplification on consonant audibility and forward masking

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Dynamic range compression (DRC) is a widely-used hearing-aid compensation strategy. The speed of gain reduction and recovery in a compressor are dictated, respectively, by its attack and release time constants. It has been hypothesized that fast-acting compression, characterized by release times shorter than 200 ms, can provide superior speech audibility and improve the rate of recovery from forward masking in hearing-impaired (HI) listeners. On the other hand, it has been reported that fast-acting compression can lead to distortions of the temporal envelope of the stimuli and degrade speech recognition. Here, the effects of DRC on HI listeners’ consonant identification in quiet and in interrupted noise were investigated. Several input levels of speech and two compression conditions were considered that differed only in terms of the release time: fast-acting (10 ms release time) and slow-acting (500 ms release time). A benefit of fast-acting compression was observed at the lowest speech input level in quiet and at medium speech levels in noise. No detrimental effects of fast-acting compression on recognition were found at any of the tested speech levels. Additionally, the two compensation strategies were evaluated in terms of objective measures such as the output gain, envelope distortion index (EDI) and a metric of consonant audibility. The average amount of temporal envelope distortion was found to be minimal, consistent with the results of the perceptual evaluation. Consonant audibility was found to account for a large part of the variance in the individual performance scores. However, the listeners seemed to differ in how efficiently they use the audible information to correctly identify the consonants. The results provide more evidence for beneficial effects of fast-acting DRC, at least in a limited class of acoustic scenarios.

D6-P-09: Influence of individual factors and acclimatization on the perception of noise reduction settings

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Hearing aid (HA) users differ substantially in their preference for noise reduction (NR) strength. This preference for and performance with NR processing are typically not correlated with each other (e.g. Neher 2014; Serman et al. 2016). In other words, HA users may prefer a certain NR setting but may perform better with another one.

The current study investigated the influence of individual noise sensitivity, HA experience and auditory acclimatization on the preference-performance relationship for different NR settings. A longitudinal study with three laboratory assessments distributed over a 12-week period was conducted. Groups of inexperienced and experienced HA users (N = 20 each) were bilaterally (re)fitted with test hearing aids in accordance with the NAL-NL1 prescription rule. Participants were selected based on an assessment of preferred NR strength performed at an initial screening visit (N = 100). An effort was made to ensure notable differences in preferred NR strength (‘NR haters vs. lovers’) in each group, and to match the two groups in terms of age and pure-tone average hearing loss. In order to investigate the effect of auditory acclimatization, half of the subjects were send home after the second laboratory assessment with increased gain while the other half remained on the initial gain prescription. A control group of experienced HA users (N = 10) completed the study with their own HAs.

Laboratory measurements were performed with four different HA settings: (1) unprocessed (amplification
only), (2) single-channel NR, (3) directional microphones (DIR), and (4) DIR combined with single-channel NR. Preference was assessed using a spatially dynamic speech-in-noise task that required the participants to attend to a target talker while ignoring two concurrent distractor talkers. Speech understanding and recall were assessed using a listening span test (Neher et al. 2018). Further data on individual factors and subjective ratings of listening effort and satisfaction during the home trials were collected by questionnaires.

In this contribution, we focus on how preference for and performance with different NR settings change with HA use and auditory acclimatization, and how HA experience and individual noise sensitivity modulate these changes. Since rather new experimental procedures were applied in this study to determine the individual performance, test-retest data were determined with the control group without intervention at the same time intervals as the experimental groups.

D6-P-10: System implementation of dereverberation method using exponential averaging with attack and release time constants for hearing aids
Kotoyo Nozaki1, Yasuhiro Okawa1, Yusuke Ikeda2, Yoh-ichi Fujisaka3, and Masahiro Sunohara3
1 Waseda University
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3 Rion Co., Ltd

It is well known that a long reverberation degrades speech intelligibility. In particular, it is difficult for people who have hearing loss to listen to speeches in reverberant environments. Listeners who have severe hearing loss have difficulties while following conversations in noisy or reverberant conditions even when speaking with only one person. Therefore, introducing the dereverberation process to the hearing aid device is considered to be effective. In recent research, dereverberation techniques have been divided into the following three classes: beamforming method, blind deconvolution method and speech enhancement method. The spectral subtraction (SS) is effective not only for noise reduction but also for dereverberation. It is one of the speech enhancement methods.

In our previous research, an SS-based dereverberation method with the blind estimation of reverberation energy was proposed. We use only single-channel speech signals based on exponential averaging with attack and release time constants. The estimation accuracy of the proposed method was greater than that of a conventional method when the reverberation time exceeded 0.6 s. Since our proposed algorithm of the blind estimation has some important parameters, the degree of dereverberation can be adjusted by changing the parameters. The effect of an important parameter, which controls how much reverberation energy is estimated to be suppressed, was investigated by evaluation with speech-to-reverberation modulation energy ratio (SRMR), which is one of the objective evaluation indices. From the results, it was suggested that reverberation energy in the high frequency bands should not be suppressed, compared to the degree in the lower frequency bands.

In this research, the previously-proposed method was implemented on a prototype of an electronic board for hearing aid device, including the parameter settings. The prototype board consists of a DSP of Analog Devices ADSP-BF706, a 24-bit analog-to-digital converter (A/D) of Analog Devices ADAU1761 which has a sampling rate of 16 kHz and two analog inputs and outputs, other peripherals including operation switches, power supply connectors and so on. The input signal is analyzed by frequency-warped filterbank. The dynamic range in each frequency band is compressed. The system latency is about 8 ms. The parameter is set optimally according to the previously-mentioned results. In addition, a listening experiment was also conducted using the prototype board to evaluate the system under the actual environment which has long reverberation. The system was suggested to be effective from some comments from not only normal listeners but also mild hearing-impaired persons.

D6-P-11: Effects of multichannel compression on spectral contrast of vowels processed by real hearing aids
Jing Shen, Varsha Rallapalli, and Pamela Souza
Northwestern University

Previous data from speech processed by hearing aid simulators have suggested that multichannel compression reduces spectral contrast of vowels (Bor et al., 2008; Amlani et al., 2011). While multichannel wide dynamic range compression is commonly implemented in modern hearing aids, there is limited evidence for whether spectral contrast of vowels is affected by multichannel compression in real hearing aid processing and how this effect is modulated by characteristics of the compressor.

This project aims to examine this effect using recordings of vowel-consonant combinations processed by real hearing aids with different numbers of channels and compression characteristics. Specifically, we ask
these questions: 1) Does number of processing channels affect spectral contrast of hearing aid processed vowels? 2) Do compressor characteristics (e.g., release time, compression ratio) influence the spectral contrast effect in multichannel compression?

Results suggest an effect of number of processing channels on spectral contrast in hearing aid processed speech. The effect is also influenced by several factors including compressor characteristics, stimuli type, and hearing aid model. Clinical implications are discussed in terms of the potential impact on vowel perception by hearing aid users. [Work supported by NIH.]

D6-P-12: Electroacoustic and behavioral evaluation of an open source audio processing platform

Daniel Rasetshwange¹, Judy Kopun¹, Ryan McCready¹, Stephen Neely¹, Marc Brennan², William Audette³, and Odile Clavier³

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

D6-P-13: Effect of the number of amplitude-compression channels and compression speed on the intelligibility of speech in noise

Marina Salorio-Corbetto¹, Thomas Baer², Michael A. Stone³, and Brian Moore¹

¹ University of Cambridge
² University of Manchester

The use of a large number of compression channels in hearing aids has potential advantages, such as the ability to compensate for variations in loudness recruitment across frequency and to provide appropriate frequency-response shaping. However, sound quality and speech intelligibility could be adversely affected due to reduction of spectral contrast (Plomp, 1988), and temporal and spectral distortion (Kates, 2010), especially when fast-acting compression is used. The objective of this study was to assess the effect of the number of channels and compression speed on speech intelligibility when the channels were used solely to implement amplitude compression, and not for frequency-response shaping. Computer-simulated hearing aids were used. The frequency-dependent insertion gain recommended by the CAM2B procedure (Moore et al., 2010) for speech with a level of 65 dB SPL was applied using a single filter before the signal was filtered into compression channels. Compression using 3, 6, 12, and 22 channels was applied subsequently. The compression speed was either fast (attack 10 ms, release 100 ms) or slow (attack 50 ms, release 3000 ms). Stimuli were IEEE sentences spoken by a male, presented in backgrounds that varied in temporal envelope (2- and 8-talker babble), and signal-to-background ratio (SBR, -3, 0, and +3 dB). Twenty adults with sensorineural hearing loss were tested. For the two-talker babble the mean scores for 3, 6, 12, and 22 channels were 55, 57, 55, and 59 RAU, respectively, for fast compression and 55, 54, 55, and 58 RAU, respectively, for slow compression. For the eight-talker babble the mean scores for 3, 6, 12, and 22 channels were 52, 54, 55, and 56 RAU, respectively, for fast compression and 55, 58, 55, and 55 RAU, respectively, for slow compression. The number of channels and compression speed had no significant effect on speech intelligibility, regardless of babble type, or SBR. [Supported by the H. B. Allen Charitable Trust.]
References:

D6-P-14: Sparsity promoting adaptive beamforming for hearing aids
Ching-Hua Lee, Gokee Sarar, Bhaskar D. Rao, and Harinath Garudadri
University of California, San Diego

Background: In hearing aid (HA) application, a common listening situation has the HA user looking at the speaker with interferences coming from other directions. The microphone array beamformer is commonly used to improve conversation between the HA user and the speaker in such circumstances. The objective of using a beamformer is to preserve the desired signal coming from the look direction while attenuating interferences coming from beside or behind the HA user. Spatial processing provided by the beamformer exploits physical separation between the desired signal and the interferences to improve speech quality and intelligibility in a noisy environment. Due to the non-stationarity of the environment, statistics of the interferences change over time. Therefore, beamformers with self-adjustment mechanism are needed for real-world HA application. An adaptive beamformer is such a system that consists of two or more microphones with filters that adapt their coefficients in response to changes in the spatial characteristics of the interferences. One popular approach is the generalized sidelobe canceller (GSC) which utilizes least mean squares (LMS) type algorithms for adjusting the adaptive filter coefficients.

Approach: In this contribution, we propose a computationally efficient and effective adaptive beamformer system for HAs. We adopt the GSC implementation of the Frost beamformer as the baseline algorithm and improve its performance by leveraging the underlying sparsity of the filter coefficients. More specifically, sparsity is promoted by incorporating the $l_1$ norm as a sparsity penalty term in the ordinary LMS optimization problem, resulting in the sparsity promoting LMS (SLMS) algorithm which has recently been shown effective for the adaptive feedback cancellation (AFC) in HAs. Effects of the parameters relevant to the sparsity degree on the interference canceling performance are investigated and evaluated by speech signals from the TIMIT database.

Results: Our current findings indicate that improvements in terms of the signal-to-interference ratio (SIR) and the hearing-aid speech quality index (HASQI) version-2 can be achieved for the GSC by utilizing the SLMS for promoting sparsity. For the scenario with the desired speech signal coming from the broadside and an interference speech signal coming from 45 degrees, with the GSC beamformer operating in the 96 kHz domain, we observed 14.2 dB of SIR and 0.22 of HASQI improvements over the system with only one microphone. Incorporating SLMS in GSC improved the performance to 17.7 dB in SIR and 0.28 in HASQI. Additional results will be included in the final presentation.

D6-P-15: Evaluation of acoustic feedback cancellation for a multi-microphone earpiece using a null-steering beamformer
Henning Schepker and Simon Doclo
University of Oldenburg

Acoustic feedback in hearing aids occurs due to the acoustic coupling between the hearing aid receiver and the hearing aid microphone(s). This results in a limitation of the maximum applicable gain and artifacts which may be perceived as howling or whistling. Although adaptive filters are often used for feedback cancellation by modeling the acoustic feedback path(s) between the receiver and the microphone(s), in practice the filter adaptation is typically biased. Furthermore, the availability of multiple microphones can be exploited to improve acoustic feedback cancellation, e.g., by adaptively removing the incoming signal in the filter adaptation. For a custom earpiece with two microphones and one receiver in the vent and a third microphone in the concha [1], several multi-microphone feedback cancellation approaches based on a fixed null-steering beamformer have been recently proposed [2,3], resulting in very good feedback cancellation performance.

The focus of this contribution is two-fold. The first objective is to present an overview of different design methods to compute the fixed null-steering beamformer, either optimizing the maximum stable gain [2] or minimizing the residual feedback power [3]. In addition, while some design methods aim at perfectly preserving the incoming signal, other methods allow
for some distortion of the incoming signal. Therefore, the second objective is to present a comparative evaluation of the different fixed null-steering beamformers in terms of added stable gain (ASG) and perceptual quality of the incoming signal in challenging acoustic conditions, e.g., a telephone receiver in close distance and changing incoming signal directions. Results show that the proposed fixed null-steering beamformers with incoming signal preservation yield a large robust ASG of up to 40dB while maintaining a high perceptual quality of the incoming signal.

References:

D6-P-16: Prototype of four-channel low-latency real-time blind source separation system for hearing aids
Masahiro Sunohara¹, Chiho Haruta¹, and Nobutaka Ono²
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Understanding speech in noisy environments is one of the crucial issues for hearing aid systems. Unfortunately, current hearing aids are often ineffective in these situations. One of the promising techniques for solving this problem is blind source separation (BSS), which is a signal processing technique to extract the desired sound source from mixtures. However, real-time BSS systems that can robustly separate several sources in real-world environments have not been realized so far because of several problems, such as a substantial calculation cost, low convergence speed, long algorithmic delay, and so forth. As a state-of-the-art approach for BSS, auxiliary-function-based independent vector analysis (AuxIVA) was proposed by Ono [1]. The computation of AuxIVA converges quickly and stably, and its online version has been presented [2]. However, frequency-domain BSS including AuxIVA could not avoid an inherent algorithmic delay due to the frame analysis. To solve this problem, the authors have proposed a low-latency scheme for real-time BSS based on online AuxIVA [3]. The proposed algorithm can significantly shorten the algorithmic delay by the time-domain implementation of demixing matrices as FIR filters and the truncation of part of their non-causal components. By performing an experimental evaluation by PC simulation, we confirmed that the proposed low-latency system with an algorithmic delay of within 10 ms worked with little performance degradation compared with that of conventional frequency-domain AuxIVA. In this presentation, we show a prototype of a four-channel low-latency real-time BSS system based on AuxIVA on a laptop computer with a two-core 2.59 GHz CPU (Intel: Core i5-4310U). The algorithm was implemented with the C++ language with real audio I/O. The total latency of the system was about 20 ms (algorithmic delay: 10 ms; hardware-dependent delay: about 10 ms). The proposed system indicates the possibility of realizing future hearing aids with better speech intelligibility in noisy environments using the BSS technique.

References:

D6-P-17: The potential of speech envelope enhancement for auditory intervention in dyslexia
Tilde Van Hirtum, Arturo Moncada-Torres, Pol Ghésquière, and Jan Wouters
KU Leuven

Growing evidence exist that dyslexia, a specific learning disorder characterized by severe reading and spelling difficulties, is related to a temporal processing deficit. A low-level dysfunction to process temporal information, might result in a subtle speech perception deficit and in turn interfere with the development of phonological representations (speech sounds representations) and literacy skills. Previous results from our ongoing work indeed reveal low-level temporal deficits, mainly with processing specific onset cues, in addition to speech perception def-
icits. Transient parts in the speech signal, such as onsets, are important for speech intelligibility in normal-hearing listeners. Therefore we hypothesize that enhancing these particular cues in the speech signal might benefit speech perception in adults with dyslexia. In the present study we implemented an Envelope Enhancement strategy (EE, Koning and Wouters, 2012), an algorithm originally developed for cochlear implant users. The EE strategy amplifies the onsets of the speech envelope without affecting other parts of the speech signal. We investigated (1) speech-in-noise abilities in a group of typical readers and adults with dyslexia and (2) the potential of the EE strategy in dyslexia research. We tested speech understanding in four different conditions: natural speech, vocoded speech and their enhanced versions. Our results show that speech enhancement instantaneously improved atypical speech perception in adults with dyslexia. Moreover, subjects with dyslexia not only benefit from envelope enhancement, but benefit more from it than typical readers. Together these results suggest that there is enormous potential for the use of signal processing algorithms for auditory intervention in dyslexia.

D6-P-18: Compensation for impaired temporal processing
Alan Wiinberg1, Morten Løve Jepsen1, Johannes Zaar2, and Torsten Dau2
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One of the most common reported complaints from people with sensorineural hearing loss is difficulties in understanding speech in complex acoustic environments. Some of these difficulties are caused by temporal processing deficits, reflecting an enhanced internal representation of the envelope signal in the auditory system and a reduced ability to discriminate envelope fluctuations in a sound. These deficits may result from loss of cochlear compression and inner hair-cell damage.

In this study, the influence of hearing loss on two measures of temporal modulation sensitivity was investigated. For modulation detection, the sensitivity was similar or higher for the hearing-impaired listeners than for normal-hearing listeners. In contrast, the modulation-depth discrimination sensitivity was lower for the hearing-impaired listeners than for normal-hearing listeners. In an attempt to compensate for the abnormal temporal processing, two non-linear amplification strategies were considered: Fast-acting compression could restore the hearing-impaired listeners’ modulation detection thresholds back towards the level observed in the normal-hearing listeners. However, the compression processing had no effect on the modulation-depth discrimination thresholds. Therefore, a temporal envelope enhancement scheme was developed can restore normal modulation-depth discrimination thresholds for modulation frequencies between 10 and 20 Hz. The performance of this enhancement scheme was evaluated by measuring consonant recognition in normal-hearing and hearing-impaired listeners using consonant-vowel nonsense syllables presented in background noise. The enhancement scheme was found to slightly improve consonant recognition scores relative to linear processing.

Saturday, August 18

SESSION D7. Hearables, Wearables, and Connectivity to Other Devices; Rehabilitative Audiology and Big Data
Session Chair: Kathryn Arehart

11:00 AM D7-O-1: Trends influencing hearing devices and real-world efficacy of a self-fit method
Andrew Sabin
Bose Corporation

The hearing assistance industry is in the midst of a transformation due to several recent developments. On the technological side, advances in connectivity, miniaturization, and machine learning
are enabling a set of increasingly sophisticated and diverse features. As this technology converges, traditional hearing aid signal processing might become a feature of a multifunctional device. Further, devices that provide hearing assistance might also become part of an interconnected network of devices each of which benefit from shared information. Other important changes are occurring on the regulatory side. Specifically, the recent passage of the Over-the-Counter Hearing Aid Act of 2017 in the United States enables new channels for distribution of hearing assistance devices directly to the consumer. Critically, such distribution requires innovation in methods that allow users to fit their own devices. While several self-fit methods have been proposed, the presentation will focus on the results of field testing with one method.

11:40 AM  D7-O-2; Blinded comparison of premium hearing aids and personal sound amplification products

_Eric Hoover¹, Karen Bell¹, Thomas Behrens², and David Eddins¹_

¹ University of South Florida
² Oticon A/S

Imminent changes to the regulation of prosthetic devices intended for persons with hearing loss are expected to increase the availability and use of low-cost, self-fit devices in the United States. The present study investigated the difference in perceptual benefit for people with mild-to-moderate hearing loss when using either a premium hearing aid (HA) or one of two high-end personal sound amplification products (PSAP) that are similar to devices that will soon be sold as prosthetics targeting hearing loss under the new regulations. A blinded comparison of sentence intelligibility in background competition and sound quality was performed using a multi-coupler head and torso simulator to present the output of the devices to a listener in a separate booth. Devices were fit according to manufacturer specifications which differed by device. The intelligibility of HINT sentences was measured using co-located speech-spectrum noise, spatially-separated speech interferers at ±45 and ±135 degrees, and a simulated multi-talker café scene. Sound quality was assessed by paired comparisons of speech, classical music, and kitchen noise. Results showed better sentence intelligibility with the HA compared to the PSAPs and no difference among PSAPs. Sound quality comparisons were not significantly different across devices but indicated a preference for the HA over the PSAPs for speech and music, and a preference for one of the PSAPs for kitchen noise likely related to lower high-frequency gain. These results indicate that speech intelligibility for listeners with mild-to-moderate losses will be improved more using a premium hearing aid over high-end consumer devices.

12:00 PM  D7-O-3: Modelling hearing aid coverage in different countries and estimating value of treatment

_Nikolai Bisgaard_

_GN Hearing_

Prevalence of hearing loss is well established in many countries. The scientific literature on this topic is abundant and the consensus is that around 16 % of the population in most countries have some degree of hearing loss. WHO states that 5 % of the population suffers from a disabling hearing loss. It is also well known that the coverage with hearing aids vary considerably from country to country although very little solid information on this topic exists. The differences can arise from economic, cultural or other factors. Data on unit sales is being monitored and published in a number of countries and for countries without public data, reasonably accurate estimates can be made, but that is still only part of the answer. Factors like bilateral fitting frequency and hearing aid replace-
ment patterns are important to know in order to translate unit sales to coverage. Since 2009, Eurotrak surveys have been conducted in many different countries and have been repeated regularly in certain countries. The EuroTrak surveys do provide estimates of coverage for each country and using data from these countries, a model linking hearing aid unit sales to coverage has been developed. Using this model, the hearing aid coverage can be estimated for any country with a reliable estimate of unit sales. This work will be explained and the results presented. Multiple models for assessing the cost of untreated hearing loss have been promoted over the years. Some are relatively conservative expressing cost of lost work effort others are relying of quality of life cost models (QUALY) and bring forward rather inflated numbers. Recently, L. Hartmann has proposed a more balanced model also including the cost of treatment. This model originally developed for France is also based on the QUALY concept, but with different assessments of loss of quality of life depending on the degree of hearing loss. The cost of treatment is also differentiated according to the degree of hearing loss. The results for France is that the benefit from fitting hearing aids is tenfold the cost of the provision thus providing a very positive incentive for developing proper hearing care. This model has been generalized and applied to the coverage data developed using unit sales and EuroTrak data. The development of the generalized model and the results will be presented.

**POSTER SESSION III**

*Paired with Sessions D6, D7, and D8*

*Saturday 9:30 AM – 11:00 AM, 8:30 PM – 10:00 PM*

Posters for Session III should be put up by 8:00 AM Saturday, August 18, and taken down after 10:00 PM Saturday, August 18. Presenters should be at their posters from 9:30 AM – 11:00 AM

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**D7-P-01: An open source platform for testing hearing aid algorithms**

William Audette¹, Odile Clavier¹, Daniel Rasetshwane², Stephen Neely², Ryan McCreery², and Marc Brennan¹

¹ Creare, LLC  
² Boys Town National Research Hospital  
³ University of Nebraska-Lincoln

The development process for hearing aid algorithms usually starts by creating algorithms on a computer and processing audio only through post-processing. Many of the hardest challenges in algorithm design, however, are only seen when the algorithms are run in real-time and worn by a real human user. Post-processing will not expose these challenges. To help algorithm designers get to real-time testing sooner, we are developing an open-source audio processing platform aimed toward hearing aid developers. We seek to make real-time audio processing more accessible to the acoustics and hearing communities. Our solution, the “Tympan”, includes both hardware and software elements to help individuals move their algorithm ideas out from their computer and into the world. Furthermore, by extending open source hardware (Teensy 3.6) and software (Arduino), a new user gets the benefit of a wide community of existing help, tutorials, and example projects – all of which is aimed at new users, not experts in programming little embedded devices. We then augment this community by providing Tympan-specific libraries and examples programs. From simple time- and frequency-domain operations to full multi-channel hearing aid algorithms, the Tympan examples illustrate and educate on how to implement your real-time audio processing ideas. Being open source, it is all available to learn from and build upon. Here we present the current functionality of the platform, as well as how to access and contribute to this exciting open source project.

**D7-P-02: Predicting the cochlear dead regions in patients with hearing loss through a machine learning based approach: A preliminary study**

Young-Soo Chang¹, Jeong-Hoon Oh², Young Sang Cho¹, Sung Hwa Hong², Yang-Sun Cho², and Il Joon Moon¹

¹ Creare, LLC  
² Boys Town National Research Hospital
Introduction: Cochlear dead region (DR) is characterized by non-functioning or poorly functioning inner hair cells (IHC) in a particular cochlear region. Although several studies have reported the reliable indicators of DR based on the detection by threshold-equalizing noise (TEN) test, it is still challenging to screen the patients who need TEN tests with clinical and audiologic test information. Methods: Five hundred and forty-five ears of 390 patients (3770 test frequencies) diagnosed with the sensorineural hearing loss (SNHL) was gathered from a period of September 2010 to May 2015. Medical records, audiology results, and TEN (HL) test were retrospectively reviewed. To analyze features associated with the classification, data on gender, age, side of the affected ear, etiologies of hearing loss, word recognition scores (WRS), and pure-tone thresholds at each frequency were collected. According to the cause of hearing loss, we categorized the patients into six groups: 1) SNHL; 2) sudden sensorineural hearing loss (SSNHL); 3) vestibular schwannoma (VS); 4) Meniere's disease (MD) [13]; 5) noise-induced hearing loss (NIHL); 6) presbycusis or age-related hearing loss (ARHL). To develop the predictive model, we performed recursive partitioning and regression for classification and built a decision tree model with rpart R package. Seventy percent of overall data (calculating as frequencies) satisfying the criteria were randomly selected and used to train the algorithm. The remaining data were used to validate the algorithm after training. Results: Among 3770 test frequencies, the overall frequency-specific prevalence of DR is 6.7%. In the decision tree model, WRS (cut-off value: 42%), pure-tone thresholds at each frequency (cut-off value: 75 dB), types of disease and frequency information are informative to detect the presence of DR. Gender and age are not associated with detecting DR. When the model accuracy was calculated with the training data, the positive predictive value was 46.2%, and the negative predictive value was 93.9%.

Conclusion: Using several clinical features, we could predict the presence of DR more precisely. To improve the prediction power of the model, it may require a more flexible model or more clinical features, such as the duration of hearing loss or a risk factor for IHCs.

Objective: The purpose of this study was to evaluate the clinical efficacy of Personal Sound Amplification Products (PSAPs) by comparing the performance of PSAPs, basic hearing aids, and premium hearing aids for mild, moderate, and moderate to severe hearing loss.

Material and methods: Twenty-nine individuals, eight with mild hearing loss, twelve with moderate hearing loss and nine with moderately severe hearing loss participated in this study. This study consisted of the Korean version of the Hearing in Noise Test, speech intelligibility in noise test, listening effort measurement using a dual task paradigm, pupillometry, and a self-rating questionnaire regarding sound quality and preference. These tests were performed in four conditions: unaided, PSAPs, basic hearing aids, and premium hearing aids.

Results: For the mild and moderate hearing loss groups, there was no statistically significant difference between PSAPs, basic hearing aids, and premium hearing aids in terms of speech perception, sound quality, and listening effort. However, seven out of twelve participants with moderate hearing loss preferred PSAPs to hearing aids. For the moderately severe hearing loss group, premium hearing aids showed better performance across all tests (p<0.05) and all seven participants preferred to use the premium hearing aids.

Conclusion: The results indicated that PSAPs may be helpful for individuals with mild to moderate hearing loss. However, if the degree of hearing loss is severe, premium hearing aids which can provide enough gain across all frequencies is needed.

D7-P-04: The wider bandwidth of the Earlens contact hearing aid leads to superior sound quality for streamed audio

Timothy Streeter, Madeline Huberth, Lindsay Prusick, Suzanne Levy, Kelly Fitz, Drew Dundas, and Elizabeth Eskridge
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Recent innovations in hearing aid connectivity have introduced a feature: Made-For-iPhone (MFi) technology allows signals (e.g., telephone, music, multimedia) to be streamed from compatible Apple devices directly to hearing aids. Unlike listening to ambient sound sources, where there is a direct acoustic path that provides low-frequency content, streamed audio sources do not have a direct acoustic path. Because the majority of acoustic hearing aids (AHAs) are fitted with an open configuration to increase comfort and avoid occlusion, sound energy for frequencies below 750 Hz is lacking, which can lead to a “thin” or “tinny” sound quality. Additionally, the effective upper limit of the audible bandwidth provided by an AHA is typically about 5000 Hz (Struck, Prusick, Hearing Review, 2017). A different type of hearing aid, a contact hearing aid (CHA), drives the eardrum directly and is able to provide effective amplification over the frequency range 125-10,000 Hz while maintaining a widely vented fitting (Gantz, et al., Otology & Neurotology, 38(3), 2017). Furthermore, the CHA can be programmed to provide low-frequency output in the streamed audio mode only, without occluding the ear canal. It is well established that increasing the effective bandwidth contributes to sound quality. In particular, Moore and Tan (J Acoust Soc Am, 114(1), 2003) showed that participants rated music with bandwidth extended to both the low and high frequencies to be more natural than for restricted bandwidth. It was hypothesized that the sound quality difference due to the differing bandwidths of AHAs and the CHA would be particularly noticeable when listening to streamed audio signals. A recent study using normal-hearing listeners confirmed this hypothesis, showing that they rated the sound quality of the CHA higher than for any of the AHAs tested, primarily due to differences observed at low frequencies (Prusick, et al., presented at the American Academy of Audiology, Nashville, 2018). The current study assessed the sound quality of streamed signals for the target population—listeners with hearing loss. Preliminary results have been found to be consistent with those from the study incorporating normal-hearing participants. Points of discussion include: the Multiple Stimuli with Hidden Reference and Anchor (MUSHRA) method and paradigm (ITU-R BS.1534-3); the choice of reference and anchors; the decision to use four hearing loss profiles for tailoring amplification, rather than using individualized amplification; and differences between acoustic open-fit and closed-fit configurations. Sound quality rating results for hearing-impaired subjects will be available at the meeting.

D7-P-05: DNN-based fitting formula for hearing aids—the minimum quantity of data for sufficient training

Chiho Haruta, Kodai Tada, Masahiro Sunohara, and Makoto Tateno
Rion Co., Ltd

The adjustment of the parameters of a hearing aid for each user is one of the important processes in hearing aid fitting. Among the processes, setting appropriate gains is an essential process that has been widely studied for decades. Many fitting formulas such as NAL-NL2 and DSL v5, have been implemented to do it. Hearing aid settings should be optimized depending not only on the user’s hearing level, but also on the user’s environment, lifestyle, intention, and so on. Therefore, a fitting system that gives appropriate parameter settings on the basis of user’s such information will be helpful in achieving more efficient fitting. We believe that machine learning techniques offer great promise in addressing these challenges. In our previous report, as the first step in addressing these challenges, we proposed a DNN-based fitting system that can create a fitting formula from actual fitting records obtained by professional fitters. The input and output layers were based on the user’s hearing level and insertion gain, respectively. The DNN was trained using the above-mentioned fitting records. 2772 fitting records of two hearing aid dispensers in Japan were used for training and the result showed that the proposed system produced fitting records closer to another set of actual fitting records of the same dispensers than any other conventional formulas. The result indicates that the proposed system can reproduce fitting records from a specific population accurately and reduce the time required for gain setting. On the other hand, when considering the practical use of the system, such a large number of fitting records may not always be available. In this study, to determine the minimum number of fitting records for sufficient training, the accuracy of the proposed system is investigated by varying the amount of training data from 100 to 10,000. For each amount of data, a fitting formula is created and two types of evaluation are conducted. For evaluation, 4,000 actual fitting records that are different from the training records are used. (1) The differences between insertion gains calculated using each formula and actual gains are investigated. The result shows that the more the training data used, the smaller the difference becomes. (2) Whether each formula shows a monotonic increase in the gain with increasing hearing loss is examined. The result shows that more than 1,000 records are necessary for training to achieve such an increase.
D7-P-06: Hearing aid users' interactions with a smart phone fine tuning app

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Hearing aid users often state the desire to have control over their device settings that go beyond adjusting loudness. Several manufacturers have released smartphone fine tuning applications; however there is little published research regarding users’ patterns and interactions with these applications.

In this study with an adjustment prototype smartphone app, the adjustments of hearing aid users in laboratory conditions and real life during home trials were evaluated. The app offered a variety of modifiers that included volume, low frequency, high frequency, noise reduction and microphone directionalit.

Twenty-two adult subjects with mild to moderate-severe sensorineural hearing loss participated in this study. All participants were experienced smartphone users. The participants were fit with Audeo B Direct RIC devices, first fit default settings were used, and aids were fine tuned only for comfort at the initial visit.

The participants were trained on the use of the app. The app consisted of the five individual modifiers mentioned above which could be adjusted independently, as well as four non-custom pre-calculated gain presets in which the volume, low and high frequency modifiers were set in a particular pattern (e.g., increased high frequencies, decreased low frequencies, etc.). The participants had the option of using one of the gain presets and making further adjustments to it, or adjusting the modifiers independently. The participants could save and label their custom presets in the program list.

The participants were instructed to create custom presets in a variety of environments, relating to their daily lives. Interviews, online surveys, logging data, laboratory tests (e.g., blindered comparisons between self-created settings and hearing aid default settings in simulated environments, speech in noise tests, etc.) were evaluated and analyzed.

The study showed that the prototype smartphone app was well received by all of the participants and they were enthusiastic about the possibility of being able to self-adjust and create an average five custom presets for unique or especially challenging environments. On average, it did not appear that personal fine tuning degraded speech performance. However, additional training or counseling may be beneficial for those that performed worse on speech testing with their own settings.

D7-P-07: Do people with (very) mild hearing loss benefit from modern hearing instruments?

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Introduction: The topic of Mild Hearing Loss (MHL) and/or Hidden Hearing Loss (HHL) is currently a controversial discussed topic. Apart from the diagnosis, it could be interesting to know to what extent modern hearing technology can be useful for people with MHL or HHL. This was the major research question of a recent review (Timmer at al. 2015) of publications in the area of MHL. One of the conclusions of the authors drew out of the review was: “…that more recent hearing-aid technologies can better address the needs of those with mild hearing impairments…”. The proposed poster will present a study addressing the question raised from Timmer et al. in 2015. Method 20 subjects with mild hearing loss were fitted with commercially available hearing systems resulting in 4 aided and 4 “unaided” conditions with the gains set to “0 dB” for better blinding the participants. As well as speech intelligibility measurements, the subjects were asked to rate their subjective perception of the aided conditions in comparison to an unaided condition, in typical realistic environments during a short walk. Additionally, In-Ear recordings were made by presenting typical critical acoustical situations. Afterwards, these recordings were presented to the subjects via headphones for complete paired comparisons. At the end, the perception threshold of high frequency phonemes of unaided and all aided conditions was determined.

Results: The results showed a clear improvement of speech perception but only in quiet for all aided conditions in comparison to the unaided condition. The data derived during the short walk showed a preference of only 2 of all aided conditions to the unaided condition in situations which were found to be critical for this kind of population. The direct comparisons indicated that only one of the aided conditions was preferred to the unaided condition in a simulated noisy Café situation. The aided conditions with more amplification at high frequency and especially with frequency lowering switched on, showed a decreased perception threshold of high frequency phonemes.
Conclusions The results of the study demonstrated that current hearing aid technology provides a clear advantage for people with MHL in situations which are relevant to them. This emphasizes the meaningfulness of a hearing aid fitting for people with a rather small hearing loss confirming the hypothesis of Timmer et al. in 2015.

Reference:

D7-P-08: Benefits and shortcomings of Direct-to-Consumer Hearing Devices (DCHD): Analysis of large secondary data from Amazon user reviews using mixed method approach

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Introduction: There is great discussion about Direct-to-consumer Hearing Devices (DCHD) and its application for adults with hearing loss. While there is much debate about the potential benefits and limitations, there is limited scientific research in this area. The current study was aimed at understanding the benefits and shortcomings of DCHDs as reported by consumers by analysing the large text corpus of secondary data generated from Amazon user reviews.

Design: Secondary data was generated by manually gathering the user feedback for 63 different DCHDs (cost range $9.95 to $379.99) in Amazon US website, which included 11,258 unique Amazon verified user reviews. The data were analyzed using both qualitative and quantitative methods. First, automated text pattern analysis (i.e., hierarchical clustering based on correspondence analysis) were performed using the open source Iramuteq software to examine the main themes emerging from the larger data corpus (100% of the data). Second, qualitative content analysis was performed on 10% of the data to look for meaningful units within the data.

Results: The cluster analysis of large data corpus resulted in 7 unique clusters, which were named as: (1) Issues related to fit and comfort (15%); (2) Friends and family recommendations (11.8%); (3) Issues related to sound quality (11.9%); (4) Listening and conversation (16.1%); (5) Positive customer service (12.1%); (6) General customer service (14.7%); and (7) Cost and affordability (17.3%). When studying the relation between cluster and rating, overrepresentation of rating 5 was noted in cluster 2 and 7, but underrepresentation of rating 5 was noted in clusters 1 and 3. Also, overrepresentation of user rating 1 was noted in clusters 3 and 6. When studying the relation between clusters and cost, overrepresentation of devices costing $0-50, $101 to 200, and $201 to 500 was noted in cluster 3, cluster 2 and cluster 7 respectively. Also, underrepresentation of devices costing $0-50 was noted in cluster 7. The qualitative content analysis resulted in two main overarching themes, which include: (a) recommended; and (2) not-recommended.

Conclusions: The study highlights the benefits and shortcomings of DCHDs which are currently in the US market. These findings relate well to the published study results of electroacoustic analysis. These findings can help clinicians to better issues related to DCHDs and advice the consumers during clinical consultations. The findings may also be of interest to hearing instrument industry from the perspective of developing products, which are developed based on users' feedback.

D7-P-09: Early results from a study of clinical data routinely collected across the VA Audiology service: patterns and pitfalls in extracting meaning from 731,209 hearing aid fittings

Graham Naylor¹, John Cannon², Lauren Dillard³, Oliver Zobay¹, and Gabrielle Saunders²
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² Veterans Affairs Rehabilitation R&D, NCRAR

Analyzing large, longitudinal samples and extensive health information contained in electronic health records (EHRs) can dramatically influence our understanding of health conditions in clinical populations. In audiology, hearing-related outcomes are complex and impacted by multiple health-related factors; therefore, an integrated approach can yield novel findings. This presentation provides a conceptual overview of such a project, examples of the authors’ experiences in undertaking it, and initial observations from the data. Statistical report and researcher perspectives are used to address questions regarding data structure, data integrity, and the utility of EHR in audiological research.

Data were extracted from two different EHR systems used by the US Department of Veterans Affairs that can be linked by a unique patient identifier. Records
were collected for individuals with at least one procedural code indicating a hearing aid fitting appointment between April 2012 and October 2014. Records include: audiological test information, survey outcome measure responses (IOI-HA), hearing aid information (style, uni/bilateral, previous experience) and indication of sustained use (subsequent battery orders), patient demographics (age, gender, etc.), and extensive codes for non-audiological diagnoses and interventions.

In this preliminary report we describe the structure of the data, including discrepancies of procedural and diagnostic code usage, statistics summarizing patient data, and analyses showing potential consequences of not considering data integrity. Taken together, these investigations provide grounds for assessing the validity of each data dimension, and thus any relationships observed. They also reveal some unexpected early insights into the behaviour of both patients and the healthcare system.

The complexities, benefits and drawbacks of using clinical EHR data for research purposes will be discussed, and functional considerations for others engaging in similar research will be outlined.

D7-P-10: Investigating the effect of hearing aid wireless connectivity technology on speech intelligibility and listening effort
Suyeon Park, Hye Yoon Seol, Yeonkyoung Park, Sung Hwa Hong, and Il Joon Moon
Samsung Medical Center

People with hearing impairment experience communication difficulties, such as watching TV and talking on their cell phones in background noise. For hearing aid users, phone conversations can be difficult due to feedback and low signal-to-noise ratio. In order to overcome these problems, wireless connectivity technology which transmits sounds from a mobile phone directly to hearing aids has been developed to increase signal-to-noise ratio. However, clinical efficacy of wireless connectivity during phone conversations in a noisy situation has not been fully evaluated. The purpose of this study is to investigate the effect of hearing aid wireless connectivity technology on speech intelligibility and listening effort. Ten adults with symmetrical bilateral moderate sensorineural hearing loss participated in the study and completed a speech intelligibility test, listening effort test using pupillometry, and sound quality evaluation in two conditions: wireless connectivity OFF and ON. In the wireless connectivity OFF condition, the participants listened to the phone through a hearing aid microphone. In the wireless connectivity ON condition, sounds from the mobile phone were transmitted directly to the hearing aids using Bluetooth. At the end of the session, participants also completed a preference survey. Resound LINX 3D hearing aids, a ReSound phone clip, and a Galaxy S7 mobile phone were used for the experiment. Hearing aids were programmed with the NAL-NL2 formula by experienced audiologists and 8-talker babble noise was presented at 75dB SPL via loudspeakers. The wireless connectivity ON condition showed significantly better speech intelligibility scores than the OFF condition (p<0.05). Participants experienced less listening effort in the wireless connectivity ON condition than in the OFF condition (p<0.05). In terms of sound quality, there was no statistically significant difference between the conditions. Lastly, eight out of ten participants reported that they would prefer to use the wireless connectivity function on the preference survey. The hearing aid wireless connectivity feature is beneficial for hearing aid users as it can significantly improve speech intelligibility and decrease listening effort for phone conversations in noisy situations.

D7-P-11: EVOTION: Big data supporting public hearing health policies
Niels Henrik Pontoppidan1, Rikke Roossing1, Jeppe Hoy Christensen1, Doris-Eva Bamiou2, Giorgos Dritsakis3, Panagiotis Katrakazas3, Dimitris Koutsouris3, George Spanoudakis4, Bin Ye5, Athanastos Bibas5, Dimitris Kikidis5, Mariola Śliwińska-Kowalska5, Louisa Murdin5, Mark Sladen, Nikos Dimakopoulos6, Dario Brdaric7, Lyubov Trenkova8, Marco Anisetti9, Apostolis Economou10, and Paschalis Papagrigoriou11
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4 The CITY University of London
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7 Guy’s and St. Thomas’s NHS Foundation Trust
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10 Pazardzhik Regional Administration
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12 Athens Medical Center
13 Empelor GmbH

EVOTION is a three-year cross-disciplinary project funded by the European Union were partners with clinical, technical, and public policy making background investigate big data supporting public hearing health policies.
EVOTION has developed a platform with components that collect usage data from their hearing aids via a smartphone, components that collect medical health records related to hearing from the clinical databases, big data analytics engine to analyse the collected data, and a new public health policy specification language for modeling of public hearing health policies. The EVOTION components employ encryption and pseudo-identifiers to preserve privacy and security.

Using the internet connection of the EVOTION mobile, the EVOTION hearing aids transmits estimators of the sound environment, and 4 bands of sound pressure level, envelope modulation parameters, and signal-to-noise ratio, to the EVOTION Data Repository one time every minute. By design, the single vector pr. minute is a non-evasive collection of sound data that preserves the privacy of communication around the hearing aids.

EVOTION has started the collection of real time usage data from more than 1000 patients' hearing aids in United Kingdom, Greece, and Denmark. Together with associated medical hearing health records the combined data allows analysis that targets individual preferences for hearing aid settings, benefits of auditory training and prevention of noise induced hearing loss in clinical settings, and prognosis of benefits from hearing aid usage to support public hearing health policies.

Preliminary analyses of the data reveal patterns of hearing aid use as function of hearing loss, sound environments, time of day, and experience with hearing aids.

D7-P-12: A sequential approach to trainable hearing aids

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Trainable hearing aids allow for parameter adjustment by a user after a prescriptive fitting by an audiologist. While potential benefits include greater patient satisfaction and device usage, existing user-driven training approaches struggle with the number and complexity of the control parameters available in modern hearing devices. Here, we describe a sequential approach to hearing aid tuning that allows a user to train a hearing aid in everyday listening situations. It relies on a machine learning algorithm to efficiently employ user inputs in navigating the space of parameters toward an optimum setting. The system relies on the ability of a human user to make reliable comparisons of sound quality under different parameter settings. We describe the results of a study that examines the ability of a small cohort of listeners to make these kinds of judgements and to effectively use the proposed training method.

D7-P-13: Hearing aid maintenance by older adults: Time course and intervention

Kimberly Skinner, Larry Humes, Dana Kinney, Sara Rogers, Annie Main, and Tera Quigley
Indiana University

Little is reported on how well older independently-living adults maintain their hearing aids, the likelihood of continued recurrence of the same problem, or the likelihood that having one problem leads to others. With the exception of annual hearing evaluations that may include hearing-aid checks, it is not common for hearing aid condition to be evaluated at regular intervals in the absence of a reported problem. Data on hearing aid maintenance is more readily available for nursing home residents and for the pediatric population. Our presentation is novel in that we report hearing aid maintenance data on a sample of healthy, active older adults. We also seek to determine the effectiveness of audioligic intervention for commonly-occurring hearing aid problems, and to identify an optimal time interval for hearing aid checks. As part of research evaluating an auditory-training program, audiologists visited participants in their homes at weekly intervals for five weeks. Participants also visited the Audiology Research Laboratory at Indiana University after that at intervals of 6 weeks, 11 weeks, 4 months, and 10 months from the initial session. At each visit, hearing aids were inspected and the condition of components of the hearing aid (e.g. tubing, dome, battery) were recorded. We report data for 57 participants aged 55-79 years wearing open-fit mini-BTE hearing aids. We found that hearing aids were well-maintained overall in the early stages of the study, and found little change during the initial five weekly visits. We found a greater number of hearing aid problems at several subsequent visits. In particular, problems were greatest for the final session, ten months after the initial session (which was also six months following the previous visit). We found the most commonly-occurring problems were with tubing, the sound bore, and the battery. We cross-tabulated the more commonly-occurring hearing aid problems and did not find that the presence of one problem correlated with the presence of another. We also cal-
culated the likelihood of a particular problem recurring for a given participant within the ten-month time frame. We had participants fill out questionnaires including a 5-point Likert-scale hearing-aid satisfaction survey and we will present correlations between maintenance scores and satisfaction scores. Our results offer support for scheduled hearing aid checks at regular intervals and also suggest optimal scheduling intervals for such checks. Our results suggest that audiolologic intervention is effective for most of the hearing aid problems observed in this study.

D7-P-14: Evaluating and improving accessibility of over-the-counter hearing assistive devices
Matthew Waggenspack, Kristi Oeding, Peggy Nelson, and Elizabeth Anderson
University of Minnesota

The number of people with untreated hearing loss is increasing. The number and types of hearing technologies available to consumers is also growing, and current legislation is changing regulations to increase accessibility and affordability of these technologies. Consumers need a resource to make sense of these hearing technologies that have similar features and functionality so they can make informed decisions when managing hearing loss. This proposal will examine over-the-counter (OTC) hearing devices using a new protocol examining objective and subjective benefit. This information will be used to create a rating system to guide consumers on which OTC will provide the greatest benefit for them. This protocol expands on previous protocols which examined electroacoustic analysis alone or one other factor. Instead, this protocol also incorporates testing of speech recognition in quiet and noise and has a real-world trial of usability and benefit. In the past, research on OTC devices was mainly disseminated to other clinicians. Information from this study will be made accessible by creating a rating system to help consumers choose the best device for them. Future projects will need to update this information as OTCs change rapidly. Models for disseminating this information to the consumer are discussed.

Saturday, August 18

SESSION D8. Evaluation in More-Realistic Environments
Session Chair: Karolina Kluk

5:00 PM  D8-O-1: Bringing the real world into the lab and vice versa: More realistic assessment of hearing ability and device benefit
Jörg Buchholz
Australian Hearing Hub, Macquarie University

Laboratory-based performance measures of speech intelligibility and hearing device benefit often do not correlate well with the performance experienced by hearing-impaired subjects in the real world. The main reasons are the rather unrealistic stimuli as well as the tasks that are commonly applied. Mainly word or sentence tests are used to assess a listener’s ability to understand speech in noise. This is very different to a real-world communication situation in which listeners need to comprehend what is said, interact with communication partners, have to maintain attention for a prolonged time, and can utilize context information. Moreover, the applied sentence material is commonly created by a trained person reading scripted sentences, which results in very clear speech that enables the development of tests with very good psychoacoustic properties but is very different from natural speech. Natural speech has a very different sentence structure, prosody, and talking speed, involves disfluencies, and talkers adapt their vocal effort and communication behavior to the given acoustic environment as well as to the hearing ability of the communication partner.
Furthermore, the background noises that are applied in the laboratory often involve only a few pre-recorded talkers, babble noise, or speech-shaped noise, which is presented from a small number of loudspeakers inside an audiometric booth. These background noises do not reflect the complexity of real-world environments, such as observed at a cocktail party or in a busy cafeteria, which include multiple interacting and moving talkers at various distances and directions, room reverberation, and diverse environmental noises such as air conditioning, fridges, coffee machines, road traffic, food steps, laughter, etc. Such realistic noises are not only different by their temporal, spectral, and spatial behavior, but can also involve distracting sources (e.g., talkers) that need to be actively suppressed by the listener while directing attention to a target talker. Related to these issues, speech intelligibility is commonly assessed at very low, often negative, signal-to-noise ratios (SNRs) that are very different from the SNRs experienced in the real world. Besides the auditory system operating differently at such low SNRs, it most likely affects the impact of a hearing loss, and signal enhancement strategies in hearing devices are evaluated at SNRs for which they have not been optimized. This presentation provides an overview over recent developments that address these limitations to improve the ecological validity of hearing outcome measures, focusing on the research conducted at the Australian Hearing Hub.

5:40 PM D8-O-2: Perceptual comparison of 3D audio reproduction system with hearing devices: A focus on localization

Laurent Simon¹, Hannes Wüthrich², Andrea Kegel¹, and Norbert Dillier¹
¹ Universität Spital Zürich
² Sonova A. G.

To design more ecological evaluation methods for hearing devices, it is possible to use 3D audio reproduction systems to simulate realistic environments. To that day, few studies considered these 3D audio systems in a hearing device context. In previous studies, the performance of a hearing device beamformer was compared objectively for various 3D audio techniques: Higher Order Ambisonics (HOA), Vector-Based Amplitude Panning (VBAP), Multiple-Directions Amplitude Panning (MDAP), and Distance-Based Amplitude Panning (DBAP). It could be verified that DBAP and third order HOA are insufficient for hearing device evaluation, but that fifth order HOA, VBAP, and MDAP could be considered for further use. In the present study, normal hearing subjects were asked to localize noise bursts produced with either fifth order HOA, VBAP, MDAP, or with real sources. They performed the task with and without hearing aids. 3D audio reproduction systems were rated in terms of time required to localize the sound sources, error of localization, and number of changes of direction of the head trajectory. Results show a significant effect of the reproduction system on the error of localization and of the processing technique on the localization time.

6:00 PM D8-O-3: What’s going on? Individualized evaluation in the real world

Inga Holube¹, Petra von Gablenz¹, Ulrik Kowalk¹, Markus Meis², and Jörg Bitzer¹
¹ Institute of Hearing Technology and Audiology, Jade University of Applied Sciences
² Hörzentrum Oldenburg

Hearing loss is a common impairment on population level. Nonetheless, hearing rehabilitation and coping essentially remain on the individual level. Since hearing loss is an individual condition, audiologists often complement audiometric profiles derived in standardized test settings with questionnaire-based disability measures addressing everyday experience. Currently used questionnaires focus on predefined listening situations and rarely consider the individual’s specific needs and circumstances. Hence, the impact of hearing impairment on the personal everyday life and the benefit
of hearing devices might not become evident. Moreover, paper-and-pencil questionnaires are usually filled out retrospectively. Therefore, the responses are potentially influenced by memory bias.

To overcome the constraints of these conventional methods, we developed assessment tools for highly individual momentary data collection in every-day listening situations. The project Individual Hearing Aid Benefit in Real Life (IHAB-RL), funded by the Hearing Industry Research Consortium (IRC), forms part of our long-term research objectives. IHAB-RL focuses on the method of Ecological Momentary Assessment (EMA) and on behavior observation to assess the health-related quality of life as relating to social inclusion, participation, and emotional well-being. For the EMA, a smartphone system with two head-worn microphones has been devised. The hardware enables the recording of objective acoustical parameters and further off-line analysis without storing any audio signals, thus preserving the privacy of all communication partners and bystanders. The objective data is used to characterize the environment in respect to levels and spectra as well as to distinguish between the system operator’s own voice and other voices or sound sources to identify communication situations. Simultaneously, an application running on a smartphone system allows for subjective ratings on predefined scales, situational descriptions using staggered queries and free descriptions, e.g., of current activities. In addition, communication situations that are important and yet challenging to the listener are more specifically explored by systematic external observations of the individual’s body postures and movements when conversing with communication partners. A code system for behavior analysis based on the International Classification of Functioning, Disability and Health (ICF) concepts activity and participation is used for this purpose. The presentation summarizes past experiences with these new assessment methods in audiology and opens up discussions on future cooperative research using a flexible open-source system and open databank concept. Everything from hardware design to software tools is and will be available on GitHub.

6:20 PM  

D8-O-4: Paired comparisons of preference for hearing-aid settings measured in the field and in the laboratory – Evaluation of LEAP, a new laboratory test based on the CoSS framework

Karolina Smeds, Josefina Larsson, Florian Wolters, Martin Dahlquist, and Petra Herrlin
ORCA Europe, Widex A/S

In a previous study, we compared two hearing-aid settings in the field and in the laboratory. We found that the correspondence between the laboratory and field data was low. Specifically, the laboratory data could not predict individual overall field preference.

In another study, we investigated the listening situations people encounter in real life. Based on a literature study, a framework called the Common Sound Scenarios (CoSS) was developed. Three intention categories were formed: “Speech communication”, “Focused listening” (without own speech), and “Non-specific” (including monitoring of surroundings and passive listening). When studying the CoSS framework, it becomes obvious that most laboratory tests only tap into the “Focused listening” intention category. Neither real speech communication, nor more passive listening situations are usually included in laboratory tests. If we want the results of a laboratory test to correspond to the results from a field study, we need to broaden the scope of our testing in the laboratory.

In the current study, 19 experienced hearing-aid users compared two hearing-aid gain settings. In the field, Ecological Momentary Assessments (EMA) were used. A smartphone was used for prompting the assessors to report on their auditory ecology and to make paired comparisons, using a remote control.

In the laboratory, a new paradigm called Live Evaluation of Auditory Preference (LEAP) was used. Six mandatory scenarios (representing all three intention categories in CoSS) and up to six individual test scenarios were included. The individual test scenarios were selected from a list of situations experienced in the field trial. Real conversations between the test participant and one or two test
leaders (guided by the main test leader) was central to the method, but ecologically valid scenarios with focused listening and scenarios including passive listening were also included.

Test participants judged the new laboratory test to be ecologically valid and motivating to participate in. When the results from the field and the laboratory were compared, the correspondence was satisfactory, even though the background sounds experienced in the laboratory were not strictly matched to those in the field, and the laboratory loudspeaker setup was basic. Test-retest reliability for the new laboratory test was evaluated and found to be satisfactory.

The new laboratory test LEAP is based on the CoSS framework and focuses on listening scenarios including a variety of intentions and tasks and on commonly experienced sounds. LEAP seems feasible to use for evaluating hearing-aid gain settings in the laboratory.

6:40 PM  

**D8-O-5: The effect of age on real-life listening**

*Annelies Devesse, Astrid van Wieringen, and Jan Wouters  
KU Leuven*

**Background:** Having a successful conversation in a noisy and dynamic environment can be challenging. Next to the auditory processing of speech, cognitive processes like attention, working memory and executive functions as well as the processing of visual (speech) cues are of particular importance during complex real-life listening situations. An increased cognitive load can result in feelings of tiredness or a lack of motivation to take part in a conversation, even though the peripheral auditory processing of sound is intact. Both cognition and perceptual processing tend to decline with age, making it even harder for older people to have a successful conversation in real-life, dynamic listening situations. In this project we investigated the effect of multitasking on speech understanding and the cognitive load during listening, i.e. listening effort, and examined the effect of aging on these individual processes.

**Method:** We used an ecologically relevant speech-in-noise test, the AVATAR-approach, to assess speech understanding abilities and cognitive load in a dynamic real-life listening scenario. Speech was uttered auditory-visually by virtual humans, displayed on a big screen. Auditory stimuli were provided through a set of six speakers. Extra auditory and cognitive tasks were added to the test paradigm to mimic real-life multitasking; in the most complex scenario, the listener executed four tasks at the same time: a speech-in-noise test, an auditory localization test, detection of direct sounds passing by and short-term memory storage of visual stimuli, imposing an extra cognitive load. Two groups of normal hearing (NH) listeners participated: a group of 35 young adults (YNH, 18-30 years old) and a group of 31 middle-agers (MANH, 45-60 years old).

**Results & Discussion:** Results showed that for both the YNH and MANH group, speech intelligibility was robust to the amount of tasks that had to be performed simultaneously. On the contrary, performance on the additional auditory and cognitive tasks dropped with increasing task complexity, indicating listening effort increased when the listening situation became more complex. This drop in performance was more prominent for the MANH group, suggesting they expended more listening effort during speech understanding in complex real-life listening scenario’s compared to the YNH group.
D8-P-01: Sound advice: An interactive learning environment for optimizing use of hearing assistive technology
Dragana Barac-Cikoja, Kevin Cole, Andrea Kottlowski, Ashleigh Collis, and Kelsey Uguccioni
Gallaudet University

An integrated suite of software applications has been developed to manage unsupervised and individualized auditory training and testing of hearing device users. It allows the users to explore simulated real-life listening situations in order to directly experience the benefits and limitations of their devices. It consists of thematic exercises that require the user to manipulate both acoustic parameters and the hearing device settings using a variety of graphic interfaces. In addition, the system tracks the user’s exploratory activities, collects his/her evaluation of the resulting listening experiences (e.g., ratings of speech intelligibility, overall sound quality, listening effort), records answers to content-related questions (e.g., what was the conversation topic or the nature of an incidental sound) and/or commentaries. All of these data are saved for later analysis. A study of the system’s usability in auditory rehabilitation (AR) from the perspective of AR providers was conducted. The participants were asked to design three exercises for a specified client. The exercises included different types of visual interface, each presenting a different task for the client: (i) manipulating the signal to noise ratio using a control panel, (ii) detecting incidental sounds and identify their source location using a graphic interface, and (iii) changing the soundscape and optimizing listening conditions by moving an avatar in a virtual space. The tasks involved listening to speech, music, or environmental sounds in noisy backgrounds typical of a restaurant, living room, office, street, or car. Objective measures of the participant’s ability to learn how to design the exercises, and their evaluation of the learning process using System Usability Scale (Brooke, 1986) and Task Load Index (Hart and Staveland, 1988) will be reported. [The contents of this presentation were developed under a grant from the National Institutes on Disability, Independent Living, and Rehabilitation Research (NIDILRR grant number 90RE5020).]

D8-P-02: Benchmarking detection and classification in automatic hearing aids
David Eddins¹, Donald Hayes², and Erol Ozmeral¹
¹ University of South Florida
² Sonova

Hearing aids constantly categorize acoustic environments into several broad classes. These classes are mapped onto distinct customizable programs. Each program represents one generic acoustic scene. This continuous, automatic classification is the basis for automatic hearing aid program switching. Clinicians separately specify gain models and adaptive features to customize performance in each available program. However, the most precise clinical optimization for each program is of no value if the classification is flawed. Classification mismatches between actual acoustic environments and the mapped programs will yield unforeseen parameterization errors. We will describe three generic tools developed to benchmark the detection and classification schemes of automatic program-switching hearing aids. Three tools will be described, referencing benchmark results from a pair of premium hearing aids.

Sound Parcour: Used to measure the precision with which hearing aids classify multiple generic listening environments. A set of 32 listening environments was created to replicate generic real-world sound classes. Those listening environments were virtualized by playback using a circular array of 24 point-source loudspeakers. The environments consisted of varying mixtures of talkers, noises, music, reverberation times, signal-to-noise ratios, signal-to-music ratios, and presentation levels.
Acoustic Azimuth Detection: Used to measure the precision of target speech detection for five conditions: no speech or speech from 0°, 90°, 180°, or 270° azimuth. Four SNR’s were tested; -3 dB, 0 dB, 3 dB and 6 dB for each speech condition. Localization accuracy was measured relative to the onset of the acoustic scene as well as fixed and variable offsets to show temporal delay of system localization. All measures were replicated in three different physical locations each with a unique set of room acoustics.

Subjective Spatial Recognition: Assesses the speed with which a spatial detection algorithm converges on a correct target and its effect on behavioral localization. Participants performed an aided localization task in quiet and in noise for speech presented from one of eight azimuths; 0°, 45°, 90°, 135°, 180°, 225°, 270° or 315°. Speech tokens were presented up to four times in a 16 second interval from each randomly selected azimuth. Participants indicated their perception of the direction of the speech token after each presentation. Our goal was to create benchmarking tools based on generic listening conditions encountered by typical hearing aid wearers. These tools provide a reasonable benchmark for comparisons of the balance between classification convergence speed and detection accuracy as the study results will demonstrate.

D8-P-03: Ecological momentary assessment: A field evaluation of subjective ratings of speech in noise as a function of acoustic and task variables

Lorienne Jenstad1, Gurjit Singh2, Anita DeLongis3, Flora Pang1, Myron Huen1, Vincent Meyer4, Rachel Ho4, and Ellen Stephenson1
1 University of British Columbia
2 Phonak Canada
3 Sonova AG

Purpose: As hearing aid outcome measures move from retrospective to momentary assessment, it is important to understand how in-the-moment contextual factors influence subjective ratings. We aimed to determine, over an 8-week period of participants responding to ecological momentary assessments (EMAs) in their own environments, 1) whether participants will complete surveys in listening situations where they are trying to understand speech; 2) whether ratings of speech in noise are similar to those obtained in the lab for similar acoustic conditions; and 3) whether EMAs tell us something different from retrospective ratings.

Methods: Fourteen adults aged 26-86 years with at least 6 months hearing aid experience were recruited for an 8-week crossover field trial (4 weeks wearing hearing aids with no EMA; 4 weeks wearing hearing aids with EMA). Participants were fitted with hearing aids and provided with a streamer and a smartphone with an app that analyzed the acoustic signal from the hearing aids and alerted the participant to respond to a questionnaire when pre-determined acoustic conditions were detected. Participants were prompted to answer questionnaires up to 9 times per day, answering brief establishing questions, then quality ratings, and finally benefit, residual activity limitation, and satisfaction. Results: Participants responded to an average of 5.5 surveys per day. Quality ratings changed predictably as the acoustic conditions changed. EMAs and retrospective ratings approached significant difference for benefit (p = .08), and were significantly different for residual activity limitation (p<.05). In both cases, EMAs were more positive than retrospective ratings. Additionally, there was large variability in EMAs across individual ratings.

Conclusions: Younger and older adults are able to complete EMAs triggered by their hearing aids. Predictable quality ratings suggest good reliability and validity of the measurement tool. High variability with EMAs may allow for nuanced detection of problematic listening situations.

D8-P-04: Temporal stability of auditory ecology of adult hearing aid users

Erik Jorgensen and Yu-Hsiang Wu
University of Iowa

Objective: Many hearing aid technologies that provide benefit in a laboratory setting do not provide benefit in the real world. One reason for this is that laboratory settings do not adequately represent the auditory ecology of the real world. Auditory ecology is multifactorial and can be defined as the interaction between physical/social context of the listening task and the listener’s perception. This context-perception interaction is dynamic and may change over time and place. As the first step to characterize auditory ecology, this study was concerned with understanding, how the contextual and perceptual factors of adult hearing aid users change over time in a given place (i.e., the “stability” of these factors over time).

Design: Fifty-four participants were fitted with bilateral hearing aids and completed ecological momentary assessments (EMA) via a smartphone app. The assessment repeatedly asked participants to report on contextual (e.g., noisiness of the environment) and
perceptual (e.g., degree of speech understanding) factors in their listening situations. A total of 11,155 surveys were included in this study. Using the GPS locations registered by smartphones during each survey, a “stability index” that represents the degree to which various contextual and perceptual factors change or remain the same in one place over time was calculated for each factor. This index was calculated by determining how often participants reported the same context or perception in the same GPS location at different times.

**Results:** The stability index differed considerably across contextual and perceptual factors. Some trends, however, were observed. Contextual factors which remained relatively stable (were the same more than 60% of the time), were level of noise and signal type (speech vs. non-speech). Contextual factors which were less stable were noise orientation, talker familiarity, talker orientation and access to visual cues. The stability indexes of perceptual factors were more homogenous and most factors (localization, loudness, loudness satisfaction, and hearing aid satisfaction) remained the same between 60% and 70% of time. Less stable perceptual factors were listening effort and speech perception.

**Conclusions:** These results suggest that the characteristics of noise are more stable than other contextual factors such as access to visual cues. These differences in stability may contribute to the relative stability of acoustic perceptual factors such as loudness and loudness satisfaction compared to speech perception and listening effort. These findings may be useful when studying hearing aid outcomes and may help to improve GPS technologies being used in hearing aids.

**D8-P-05: An open source toolkit for privacy-preserving real-world EMA data collection**

**Ulrik Kowalk, Inga Holube, Sven Franz, Holger Groenewold, Petra von Gablenz, Sven Kissner, and Jörg Bitzer**  
Institute of Hearing Technology and Audiology, Jade University of Applied Sciences

Modern audiological studies show a deep interest in the combination of objective data and subjective ratings, especially when personal benefit of rehabilitation measures is the desired entity. In order to perform experiments that gather data of both varieties in an ecological, compliant, and privacy-preserving manner, we developed and evaluated a system modeled around a smartphone allowing for comfortable usage with minimal user interaction needed. Audio signals recorded close to the ears are transmitted to the device via a stereo A2DP bluetooth stream. All objective parameters are then extracted in real time and no audio data is stored. Simultaneous to recording objective data, surveys are taken in situ on the smartphone at a fixed or random time interval. Questionnaire design follows a simple set of rules and new surveys can be produced without any programming skills. This system is the central element of data collection within the IHAB-RL study on hearing aid benefit in real-world scenarios.

The technical assessment of the system has shown a flat frequency response with low inherent noise, a sufficient dynamic range, and a runtime suitable for real-life experiment purposes. The interface itself has been designed for unambiguity and clean display of only the most essential variables. While the software may traverse different states and handle various tasks in the background, the user is only involved in case of critical battery level or permanent loss of the down-link. With our new evaluation software, we are able to display compressed results of whole multi-day experiments (compare the corresponding IHCON contribution Gablenz et al, 2018). An external module checks all data blocks for plausibility, ensuring that only meaningful data is included. In order to manage the huge amount of data and enable international collaboration, we designed and implemented a database, which is specifically structured for ease of access, expandability, and assignment of individual permissions. Access and permission management is facilitated by a specially tailored programming interface (API).

Furthermore, a revised hardware design is presented which utilizes a serial wireless connection to the benefit of increased dynamic range, broader control of the system variables and a less restricted catalog of usable hardware. In order to promote collaborative approaches, all software is published under an open source license while all hardware plans are made available under an open hardware license.

**D8-P-06: Tracking eye and head movements in natural conversational settings: Effects of age, hearing loss and background noise level**

**Hao Lu\(^1\), Martin McKinney\(^2\), Tao Zhang\(^2\), and Andrew Oxenhaim\(^3\)**  
\(^1\) University of Minnesota  
\(^2\) Starkey Hearing Technologies

Beamforming algorithms for hearing aids typically assume that the listener is directly facing the target talker for simplicity. However, the listener’s head
may not always be facing the target talker in natural conversation settings, potentially limiting the benefits of the beamforming algorithms in real-world environments. The aim of this study was to determine the extent to which the tracking of eye gaze improves the tracking of the target talker position in natural conversational settings, and to test the hypothesis that eye movements are more likely to track the target talker position at high background noise levels, particularly in older listeners and listeners with hearing loss. Three groups of participants were recruited: younger listeners (aged 20 to 24 years) with clinically normal hearing, older listeners (aged 60 to 76 years) with clinically normal hearing, and older listeners (aged 53 to 78 years) with mild-to-moderate hearing loss. All participants with hearing loss were bilateral hearing aid users and wore their own hearing aids during the experiment. Each participant engaged in natural conversation with two confederates, approximately equally spaced around a small table in a large sound treated booth. During the 30-minute experiment, the participant and the confederates conversed without any explicit instructions regarding topic. Different levels of background noise were introduced by playing background cafeteria noise via a surrounding loudspeaker array. The participants’ head and eye movements were tracked using a Tobii Pro Glasses 2 system. In general, the participants’ head movements tended to undershoot the position of the target talker, but head and eye movements together generally predicted the target talker position well. No significant differences in head and eye movement behavior were observed between the different listener groups, or at the different noise levels. Hearing-impaired participants tended to spend a longer time talking than the participants in the other two groups at lower levels of background noise. Overall, the results confirm that tracking head and eye movements together generally provide more accurate estimates of the target talker position than tracking head movements alone. However, the data did not reveal any significant changes in behavior as a function of background noise level or as a function of age or hearing loss. [Supported by Starkey Hearing Technologies.]

D8-P-07: Research design and pre-post evaluation of hearing aid provision under field and laboratory-controlled conditions: Contributions for postmarket surveillance requirements

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Regulatory affairs are a very important topic for medical device companies. Regulatory requirements concern both pre-launch clinical studies, and Postmarket-Surveillance (PMS) requirements. In the past, PMS was covered with reactive study designs. However, the requirements for PMS studies are changing. According to the new European Medical Device Regulation (EU, 2017/745), manufacturers are obligated to institute a systematic procedure to proactively collect post-market data. The FDA specified PMS designs (guidance 522) and proposed to gather data with real-world clinical practice, and to address device-related public health questions. The purpose of the present study was to propose a research design, which is suitable to meet the requirements of the new regulatory requirements. The basic idea was to run a prospective cohort study to test hearing aids (HA, Unitron Moxi Fit) before and after HA provision with experienced HA users (EXU) and first-time users (FTU). A field study and a controlled laboratory study were run in parallel. Both studies were coordinated and monitored by Hörzentruman Oldenburg (HZO), a centre experienced in scientific research, to safeguard quality. The field study was designed to be easy to conduct for hearing aid dispensers and patients in clinics without prior scientific research experience. 158 patients of Vitakustik shops across Germany participated in the field study (N=88 FTU, mean age=68.0 years; N=70 EXU, mean age=71.4 years). The laboratory study had a sample of N=20 FTU (mean age: 70.5 yrs.) and was conducted at the HZO. Both samples did speech in noise tests and questionnaires, i.e., SSQ, listening effort, items of the IOI-HA, and EMO-CHEQ (see contribution Singh et al., in press, Ear and Hearing). Furthermore, data-logging from the hearing aids over a few-week period was examined regarding listening environments. The FTU from both study streams benefitted significantly from HA provision in most of the tested sections and showed similar patterns regarding the scene classifier. Analyses showed that resilient data were obtained in both the laboratory and clinics. Additionally, the monitoring of the clinics and the test procedures revealed good measurement standardization and high quality of the data. One challenge that should be addressed in future research concerns the increased missing data observed in the field relative to the laboratory. In sum, the current research design appears as a viable way to meet regulatory requirements of the EU and the FDA for PMS studies, though clear implementation rules for PMS are still missing for the EU.
**D8-P-08: Effect of task difficulty and signal processing on recall performance and recall strategy**

*Andreea Micula1,2, Elaine Ng1, Jerker Rönnberg2, and Fares El-Azm1*

1 Oticon A/S
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Despite technological advances aimed at compensating for hearing loss in challenging listening situations, many hearing aid users are not satisfied with their hearing aids. Complaints may arise even in listening situations where aided speech understanding is good. These may result from problems other than audibility, such as high degree of listening effort. Current clinical outcome tools (e.g. audiometry, speech audiometry) mainly measure audibility and do not consider other factors involved in daily communication (e.g. processing and remembering speech while preparing a response). The Sentence-final Word Identification and Recall (SWIR) test (Ng et al. 2013) is an auditory recall test designed to investigate the effect of hearing aid signal processing on memory for highly intelligible speech in background noise. The task is to repeat every last word of a sentence in a list and recall the last words when the list is finished. Previous findings suggest that people with high cognitive capacity benefited more from advanced hearing aid signal processing than people with low cognitive capacity. However, people with low cognitive capacity showed benefit when a less demanding version of the test was used. Thus, excessive task difficulty could have prevented capturing any potential benefit from hearing aid signal processing. Other findings suggest that people with low cognitive capacity may benefit more from milder forms of signal processing with less sound distortion. The purpose of this study is to continue developing the SWIR test by manipulating task difficulty. This is done by varying the list length and thus the number of final words to recall. We will also compare whether knowing the total number of final words before a list (task difficult predictability) affects recall performance and recall strategy. The effect of noise (four-talker babble and speech-shaped noise) and noise reduction on recall will be investigated. Forty native Swedish speakers aged 40 – 70 years are recruited. All participants have moderate to moderately-severe adult-onset symmetrical sensorineural hearing loss and minimum one year of hearing aid experience. Besides the modified SWIR test, additional tests are administered. The Hearing in Noise Test is used to measure speech intelligibility. Several cognitive tests are administered to measure cognitive capacity and assess various areas of cognitive function. Preliminary results of this study will be presented.

**D8-P-09: The spatial speech test for assessing binaural hearing**

*Bhavisha Parmar1, Debi Vickers2, Mana Ahnood1, and Jenny Bizley3*

1 UCL Ear Institute
2 University College London

The fitting of binaural cochlear implants and hearing aids are now routine interventions for bilateral hearing loss. However, there is still a great deal of clinical concern about the approaches used for fitting and validation of binaural devices and there is a need to develop an assessment measure that provides meaningful, ecologically valid information on binaural processing ability that is sensitive to small modifications in fitting. An assessment that not only uses speech as a target stimulus but also assesses spatial speech perception at the same time, without the need for a large spatial release from masking (SRM) test battery, would be ideal.

The spatial speech test was developed to simultaneously assess the ability of listeners to identify speech and determine the relative location of target sound in the presence of speech babble (Bizley et al., 2015). During the test, listeners hear two sequentially presented words from adjacent speakers with a 15° separation and they identify the word and the relative direction of the 2nd word presentation relative to the 1st. The task is performed in the presence of multiple independent noise sources at an individually determined signal-to-noise ratio.

While this test provided a sensitive assessment of spatial hearing in normal hearing listeners, when bilateral hearing aid users were tested in the same task they were unable to perform the relative-localization aspect despite doing well on a standard clinical localization task measuring the ability to point to a speech source in silence.

The test was adapted further to make it easier for hearing impaired listeners by restricting the noise sources to left or right space and widening speaker separations to 30°. We tested normal hearing listeners (n=11) and bilateral cochlear implant users from 8-80 years of age (n=10).

The findings showed that the spatial location of the words had a significant effect on relative localization performance for both normal hearing and cochlear
implanted listeners. For the normal hearing listeners’ performance on the word identification aspect of the test was moderated by spatial separation of the words from the noise sources, this was not the case for the bilateral CI users.

Here, we present new data from bilateral hearing aid users trialling the refined spatial speech test.

Reference:

D8-P-10: The effects of microphone technology on word recognition and behavioral listening effort in school-aged children
Erin Picou1, Samantha Gustafson2, and Todd Ricketts1
1 Vanderbilt University Medical Center
2 University of North Carolina at Chapel Hill

Classrooms are notoriously difficult listening environments and children with hearing loss are even more vulnerable to the effects of poor acoustics than students with normal hearing. Advanced hearing aid microphone technologies, specifically remote microphone systems and directional microphones, have been shown to significantly improve speech recognition in classroom environments. While speech recognition benefits are clearly paramount, other factors including the cognitive resources or “listening effort” that children must exert also can greatly affect the total listening experience. For adults with hearing loss, directional microphones in hearing aids have been shown to improve listening effort in moderately reverberant laboratory environments. Based on the considerable improvements in signal-to-noise ratio with remote microphone systems, it would also be expected that they would also improve listening effort. Early work in adults and children investigating benefits of advanced microphone technologies is commonly limited by including only listening situations designed to optimize microphone benefits, for example a single talker positioned in the front hemifield. However, school-aged children are often tasked with listening and learning in dynamic environments where they are surrounded by potential talkers of interest. The purpose of this study was to evaluate the effect of microphone setting on word recognition and behavioral listening effort in school-aged children with bilateral, sensorineural hearing loss. Loudspeakers presented speech stimuli for the front and also from behind a participant. Background noise speakers presented multi-talker babble surrounding a participant. The signal-to-noise ratio was +5 dB (speech: 65 dB SPL, noise: 60 dB SPL). A dual-task paradigm was used to simultaneously measure word recognition performance and listening effort. Participants completed testing in three microphone conditions: 1) omnidirectional, 2) directional, and 3) remote microphone. Data were analyzed using repeated measures analysis of variance with two within-participant factors (loudspeaker location, microphone setting). Results revealed loudspeaker-specific effects. Advanced microphone technologies provided small word recognition and listening effort benefits when speech was in the front, but moderate-to-large deficits when speech originated from behind a participant. These results demonstrate the importance of optimizing microphone technologies for each listening situation, as the benefits of advanced microphone technologies could be outweighed by their limitations in some listening situations.

D8-P-11: An update of the Connected Speech Test
Hasan Saleh1, Paula Folkeard2, Ewan Macpherson1, and Susan Scollie1
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2 National Centre for Audiology-Western University
3 University of Western Ontario

The Connected Speech Test (CST) is a speech in babble test that allows measurement of speech intelligibility while listening to conversationally produced speech against a background of multi-talker babble. It was developed to assess the benefit of using hearing aids in challenging everyday speech environments (1). Connected speech sentences, where the sentences are meaningfully related, contain intelligibility cues which isolated word and unconnected speech sentences (no specific topic) do not. The linguistic cues, along with the addition of relative duration of phonemic characteristics within each sentence, can be used by a listener to better discern word meanings (1). Complete sentences are also a suitable representation of real world everyday speech, making them a more valid test for speech intelligibility than isolated word tests (1).

The CST consists of 24 passage pairs, with each passage consisting of 10 sentences. These are presented against a background of multi-talker babble, and the listeners are asked to repeat what they hear from the passage. The topics of these passages are given to the listeners prior to presentation, with the sentences of each passage being related to an overall topic. A score
is assigned based on how many of the 25 keywords of each passage pair are repeated.

The aim of this project is to update the CST, by recreating it using current recording equipment to improve sound quality and including a more neutral North American accented speaker for the speech stimuli. New stimuli were recorded using the same passages as the original CST, and new babble tracks were also recorded. All recordings were made with an AKG C4000 B microphone in a double walled sound booth. A female speaker was used for the stimuli recordings and the six-talker babble consisted of 3 males and 3 females reading 5 of the same passages, which were then merged to one babble track. The passages were administered to 19 (11 females and 8 males) normal hearing university students aged between 18 and 25 years of age at a speech-noise-ratio of -3 dB to the babble track, following the recommended procedure of the original CST. Analysis of the passage score distribution across the participants is in progress, and this will be used to assess the passage equivalence of the new recordings.

Reference:

**D8-P-12: Real-time simulations of hearing loss and auditory prostheses**

*Eric Tarr, Braden Carey, and Hannah Wright*  
*Belmont University*

Research to improve the treatment of hearing loss is predicated, in part, on measuring listener performance under conditions of simulated hearing loss. Acoustic hearing loss can be simulated using signal processing to replicate measured audiograms and various functions of hearing aids. Cochlear implants can be simulated using a multi-band vocoder with various types of carrier signals. These different simulations were implemented in a system to simulate hearing loss using real-time processing on a mobile device. Versions of the app are available for the iOS platform. The app was created as part of the Sennheiser AMBEO Developers Program to utilize the AMBEO Smart Headset. This headset has stereo input, stereo output, and a head-tracking device to make possible many configurations of binaural listening and three-dimensional sound. Implemented features of the app include multi-band compression, frequency smearing, and temporal distortion to replicate perception using hearing aids. A listening experiment of speech in noise was conducted to demonstrate the basic functionality of the app. The purpose of creating this system is to make possible the evaluation of listener performance in many different listening environments. Furthermore, this system provides a platform to assess additional signal processing methods for auditory research.

**D8-P-13: Data analysis from real-world hearing assessment**

*Petra von Gablenz, Inga Holube, Ulrik Kowalk, Sascha Bilert, Markus Meis, and Jörg Bitzer*  
*1 Intitute of Hearing Technology and Audiology, Jade University of Applied Sciences  
2 Hörzentrum Oldenburg GmbH*

Ecological Momentary Assessment (EMA) is a promising approach for evaluating the impact of hearing loss and the benefit of rehabilitative interventions in real-world settings. However, largely abandoning controlled test conditions involves challenges of many kinds. It certainly underlines the need for a conceptual and procedural framework that guides both the technical validation and the patient-centered interpretation of the gathered data.

In an ongoing EMA field study, a smartphone-based device is used to collect subjective listening assessments by means of an app-based digital questionnaire as well as objective acoustical parameters (see IH-CON-contribution Kowalk et al, 2018). Audio signals are received by means of two microphones mounted on the left and right temple of the participant’s glasses. However, the storage of audio features is limited to RMS levels, smoothed Auto and Cross-Power Spectral Densities (A/C-PSD) and Zero-Crossing Rates (ZCR) to ensure privacy.

Initially, an ad-hoc analysis is performed when the study participants return the EMA device, basically to inform the supervisor about the hardware runtime, data storage and validity, and the main assessment results sorted by the participants’ activities and locations. Descriptive statistics, e.g. on the frequency of assessments, mean ratings of speech understanding or listening effort are mostly graphically displayed, thus easy to grasp for the supervisor and – complemented by some oral explanations – mostly accessible to the study participant. As this EMA study is primarily designed for rehabilitative purposes, we reject to regard study participants as mere data providers, but try to let them participate as much as possible in their EMA results. In addition, this ad-hoc analysis includes so-called day prints. These figures show RMS levels,
PSD and assessments versus time, thus carry highly condensed and detailed information useful both to the supervisor and for later data analyses. Subsequently, the real part of the coherence calculated from PSDs is used to identify data segments containing the study participants’ own voice. Whatever approach is applied in order to estimate the level of signals and background noise, reliable detection of own-voice segments is crucial to prevent biased results. Reference on own-voice activity, moreover, allows for cross-validating communication situations reported in the questionnaire.

This contribution outlines this two-step data analysis established to cope with EMA data in rehabilitative hearing research and shows exemplary results derived in hearing-impaired persons.
### Attendees

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