Virtual Conference on Computational Audiology (VCCA)

25-26 June, 2026

Bridging auditory science: From neural mechanisms to clinical practice

Program & Abstracts

Hosted by

UNIVERSIDAD DE GRANADA

Computational Audiology Network

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Welcome to VCCA 2026

Dear participants of VCCA 2026,

It is our great pleasure to welcome you to the **7th Virtual Conference on Computational Audiology (VCCA 2026)**, hosted this year by the University of Granada, the University Hospital Virgen Macarena, and the University of Salamanca, in collaboration with the Computational Audiology Network (CAN).

Since its inception, the VCCA series has sought to bring together researchers, clinicians, engineers, industry professionals, students, and technology developers interested in understanding how computational approaches can transform hearing science and hearing healthcare. Across the years, the conference has become an international forum for exchanging ideas, fostering collaborations, and exploring how advances in signal processing, machine learning, neuroscience, modelling, digital health, and clinical audiology can contribute to better outcomes for people with hearing difficulties.

The VCCA 2026 programme reflects this multidisciplinary spirit. Over two conference days, participants will explore topics spanning computational audiology, binaural hearing, speech-in-noise perception, cognition and listening effort, hidden hearing loss, objective biomarkers, machine learning and artificial intelligence, tele-audiology, hearing aids, cochlear implants, translational hearing research, and multisensory integration. We are particularly pleased by the breadth of expertise represented across the programme and by the strong balance achieved between fundamental science, technological innovation, and clinical application.

This year, we are also delighted to introduce three **pre-conference workshops** designed to provide accessible, practical introductions to essential skills and concepts increasingly relevant to modern hearing research. These workshops will offer first hands-on exposure to **effective scientific communication**, **machine learning fundamentals**, and **computational modelling**, helping participants from diverse backgrounds develop new tools and perspectives that can be applied in their own research and professional practice.

We are honoured to welcome a distinguished group of **Keynote Lecturers** and **Invited Speakers** who will share their expertise, experiences, and perspectives with the VCCA community, continuing a tradition that has been central to the conference series since its inception. Their contributions span topics ranging from auditory neuroplasticity and cochlear synaptopathy to paediatric auditory physiology, hearing rehabilitation, neurofeedback, speech perception, and translational hearing research. We are deeply grateful for their willingness to share their expertise and inspire the VCCA community.

We are equally excited to recognise excellence within our own community through three conference awards: the **Best Podium Award**, the **Best Poster Award**, and the **Emerging Scientist Award**. In particular, the Emerging Scientist Award reflects our strong commitment to supporting and promoting the next generation of researchers, whose creativity, energy, and vision will help define the future of computational audiology.

The success of VCCA 2026 would not have been possible without the dedication and generosity of many individuals and organisations. We would like to thank our keynote and invited speakers, authors, workshop instructors, session chairs, technical chairs, sponsors, members of the Scientific Committee, and colleagues from the Computational Audiology Network who have contributed their time, expertise, and enthusiasm to make this conference possible.

We are also sincerely grateful to our sponsors for their trust and support. Their contribution plays a fundamental role in enabling us to provide a high-quality, fully virtual, and freely accessible international conference for the global hearing science community.

Finally, it has been a genuine privilege for us to work alongside the CAN Board, the Scientific Committee, and the many colleagues who have helped shape this programme. We hope that VCCA 2026 will stimulate new ideas, spark collaborations, support emerging talent, and strengthen the international community working at the intersection of hearing science and computational methods.

Whether you are joining us from Europe, the Americas, Asia-Pacific, Africa, or elsewhere, we warmly welcome you to VCCA 2026.

We hope you enjoy the conference.

Joaquín T. Valderrama-Valenzuela (University of Granada)
María Amparo Callejón-Leblic (University Hospital Virgen Macarena)
Miriam Isabel Marrufo-Pérez (University of Salamanca)

VCCA 2026 Organising Committee

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We are proud to acknowledge the organisations whose support has helped make VCCA 2026 possible. We encourage attendees to learn more about their work and their contributions to hearing research, hearing healthcare, and innovation by visiting their websites.

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Program

Day 1 (Thursday 25th June, 2026)

Time UTC+2	Room A	Room B
12:00 – 12:15	**Opening ceremony** Welcome. Prof. Gabriel Maciá Fernández, Vice-Rector for Digital Transformation, University of Granada, Granada, Spain. Dr. Jan-Willem Wasmann, President of the Computational Audiology Network. Dr. Joaquin T. Valderrama, Dr. María Amparo Callejón Leblic, Dr. Miriam I. Marrufo-Pérez, VCCA 2026 chairs.	
12:15 – 13:00	**Keynote Lecture 1** *"Separating the Causes of Listening Difficulties when Thresholds are Normal: Concepts and Data"* **Prof. Harvey Dillon** (Macquarie University, Sydney, Australia) *Presented by Dr. Jan-Willem Wasmann*	
13:00 – 13:10	Mini-Break	
13:10 – 14:25	**Session 1.A. Computational Audiology & Digital Hearing Research** *Model-based approaches, auditory simulations, and data-driven insights into hearing processes.* **Chaired by Dr. Helia Relaño Iborra.**	**Session 1.B. Binaural Hearing & Spatial Sound Perception** *Localization, spatial cues, and listening in complex acoustic environments.* **Chaired by Dr. Jaime Andres Undurraga Lucero.**
	Featured. "The Effect of Hearing Loss on the Characteristics of Daily Life Conversations," by Carlota Sabaté Cao.	***Featured***. "Contralateral sound attenuation can help hearing-aid users understand speech in realistic, dynamic "cocktail party" listening situations." by Paula García Zaballos.

	Podium. "Auditory Models for Hearing-loss Compensation – some Candidates," by Lars Bramsløw.	***Podium***. "Longitudinal Training Enhances Interaural Timing Sensitivity in Cochlear Implant Users," by Raymond Goldsworthy.
	Podium. "Joint Estimation of Attentional and Masking Error Rates for Speech Recognition in Competing Speech," by Adam Bosen.	***Podium***. "Headphone-based Spatial Perception Tests in Listeners with Sensorineural Hearing Loss: Insights from Receiver Operating Curve (ROC) diagnostics," by P Prameela.
	Podium. "Measurement-Aware Registry Design for Cochlear Implant Outcomes: Lessons from a 25-Year Clinical Dataset," by Apurv Shukla.	***Podium***. "Binaural Neural Processing in children with and without a history of otitis media," by Laura Hansen.
	Podium. "Software-Assisted Analysis and Classification of Cortical Auditory Evoked Potentials," by Manuel Lazo-Maestre.	***Podium***. "Speech and Spatial Feature Encoding in Binaural, Naturalistic Audio Representation Models," by Meike Span.
	Podium. "Listener-Aware Speech Representations for Hearing-Aid Applications," by Sarthak Mangla	***Podium***. "Perceptual Weighting of Auditory Localization Cues Across Frequency Bands in Free-field," by Nele Naumann.
14:25 – 16:00	Lunch / Dinner Break	
16:00 – 16:30	**Invited Speaker 1.** *"A Novel Gamified Neurofeedback System for Enhancing Attention in Naturalistic Listening Environments in Normal-Hearing and Cochlear Implant Users"* **A/Prof. Andrew Dimitrijevic** (University of Toronto, Toronto Canada) *Presented by Dr. María Amparo Callejón Leblic*	
16:30 – 16:40	Mini-Break	
16:40 – 17:55	**Session 2.A. Speech-in-Noise and Auditory Scene Analysis** *Perceptual and neural mechanisms supporting speech understanding in real-world conditions.*	**Session 2.B. Cognition, Listening Effort & Brain Connectivity** *Neuro-cognitive models, EEG/fMRI correlates, and individual differences in listening effort.* Chaired by Dr. Yue Zhang

	Chaired by Dr. Miriam I. Marrufo-Pérez.	
	Featured. "Attention to Speech Modulates Distortion Product Otoacoustic Emissions Evoked by Speech-Derived Stimuli in Humans," by Janna Steinebach.	***Featured***. "Decoding of Speech Acoustics from EEG: Going Beyond the Amplitude Envelope," by Alexis MacIntyre.
	Podium. "Speech-in-Noise Difficulties in Aminoglycoside Ototoxicity Reflects Combined Afferent and Efferent Dysfunction," by Lina Motlagh Zadeh.	***Podium***. "Auditory Processing Differences in Stuttering: Evidence from Load Theory of Attention," by Fjorda Kazazi.
	Podium. "Behavioural and Neurophysiological Correlates of Adaptation to Noise in AM Detection," by Iñigo Santamaría Fernández.	***Podium***. "Effect of Reverberation and Signal to Noise Ratio on Listening Effort in Native and Non-native Kannada Speakers," by Shivani Sharma.
	Podium. "The Effect of Changing Talker Voices on Speech Perception in Individuals with Normal Hearing," by Saranya Mundayoor.	***Podium***. "HD-fNIRS Assessment of Listening Effort in CycleGAN-Enhanced Virtual Acoustics," by Ali Syed.
	Podium. "Speech Perception in Noise in Individuals with Misophonia: Evidence from the Audible Contrast Threshold Test," by Hamssika Sudhakar.	***Podium***. "Infants Behave Like Adults During Speech-in-Noise Processing—but What Happens in the Brain?" by Irene Arrieta.
	Podium. "A Comparison of Speech Recognition Between Humans and Modern Machine Systems," by Annika Magaro.	
17:55 – 18:15	Break	
18:15 – 19:30	**Session 3.A. Hidden Hearing Loss and Subclinical Auditory Disorders** *From synaptopathy to auditory fatigue: mechanisms, models, and diagnostic markers.* Chaired by Dr. Joaquin T. Valderrama.	**Session 3.B. Objective Measures and Biomarkers in Hearing Research** *Electrophysiological, pupillometric, and behavioural indices for assessment and prognosis.* Chaired by Dr. Andrew (Andy) J. Beynon.
	Featured. "Efficient Coding Explains Altered Neural Representations in Hidden Hearing Loss," by Juan M. Fuentes.	***Featured***. "Multichannel Auditory Cortical Evoked Potentials and Source Localization Analyses in Misophonia: A Neurophysiological

		Investigation," by Kamalakannan Karupaiah.
	Podium. "Objective SNR Benefits from Hearing Aids Fitted with the NAL-NL3 MHL Module: Effects of Performance Level and Dome Type," by Gilles Courtois.	***Podium***. "Measurement-Aware Registry Design for Cochlear Implant Outcomes: Lessons from a 25-Year Clinical Dataset," by Apurv Shukla.
	Podium. "Phenotypic Changes of Auditory Nerve Fibers After Excitotoxicity," by Jérôme Bourien.	***Podium***. "Triggered: Using Computer Vision to Measure Sound Intolerance," by Samuel Smith.
	Podium. "Subclinical Effects of Recreational Noise Exposure: A Pre-registered Longitudinal Study," by Chris Plack.	***Podium***. "Altered Central Auditory Gain in Misophonia: N1–P2 Slope and Categorical Loudness Findings," by Ishita Marwaha.
	Podium. "Extended High-Frequency Hearing Predicts Rapid Spoken-Word Recognition," by Tugba Lulaci.	***Podium***. "Thalamocortical Auditory Processing in Misophonia: Evidence from Middle Latency Auditory Evoked Potentials," by Santhosh Periyasamy.
	Podium. "Deep Learning–Based Classification of Hidden Cochlear Pathologies based on ABR Waterfall Plots," by Prasad Darveshi.	***Podium***. "Otoacoustic Emission Screening Extended to the Assessment of Auditory Neural Health," by Francois Deloche.
19:30 – 19:40	Mini-Break	
19:40 – 20:10	**Invited Speaker 2.** "*Hidden Hearing Loss: From Animal Models to Human Translation*" **Prof. Sharon Kujawa** (Harvard Medical School, Massachusetts Eye and Ear, Boston, USA) *Presented by Dr. Joaquin T. Valderrama*	
20:10 – 21:00	Lunch / Dinner Break	
21:00 – 21:45	**Keynote Lecture 2.** "*Hijacking and helping auditory attention: How interruptions and prediction shape everyday communication*" **Prof. Barbara Shinn-Cunningham** (Carnegie Mellon University, Pittsburgh, PA, USA)	

	Presented by Dr. Miriam I. Marrufo-Pérez	
21:45 – 21:55	Mini-Break	
21:55 – 22:30	**Session PB1.** Poster Blitz Presentations from Sessions 1A, 2A and 3A. *Chaired by Dr. Joaquin T. Valderrama*	**Session PB2.** Poster Blitz Presentations from Sessions 1B, 2B and 3B. *Chaired by Dr. Miriam I. Marrufo-Pérez*
	Poster [1A]. "R-based Systematization of Advanced Audiometric Formulas: Beyond the 25 dB Threshold," by Ramón Hernández-Villoria.	***Poster*** [1B]. "Effect of Binaural Cue Distortions on Music Quality Ratings with Simulated Hearing Loss and Hearing Aids," by Jack Webb.
	Poster [1A]. "Audiometric Data Reproducibility with Differential Privacy Using the Laplace Mechanism," by Hector Gabriel Corrale de Matos.	***Poster*** [1B]. "Task-Optimized Models of Uncertainty Reproduce Human Confidence Judgments," by Lakshmi Narasimhan Govindarajan.
	Poster [1A]. "Re-examining Auditory Filter Parameters: A Characteristics-Based Framework with Updated Estimates for Humans," by Samiya Alkhairy.	***Poster*** [1B]. "Impact of Misophonic Triggers on Binaural Auditory Integration: Evidence from Dichotic Listening Testing," by Nesmah Parackodan.
	Poster [1A]. "Capturing Real-World Listening Experiences with Smartwatch-Based Ecological Momentary Assessment," by Nicky Chong-White.	***Poster*** [1B]. "How Does Action-Type Computer Game Playing Affect Hearing?" by Dilara Usta.
	Poster [1A]. "A Neural Mass Model of the Auditory Corticothalamic Circuitry," by Spencer Olson.	***Poster*** [1B]. "Influence of Spatial Hearing on Auditory Parallel Subitizing Efficiency," by Emil Zawistowski.
	Poster [1A]. "Student-Led Digital Innovation in Audiology: Virtual Platforms, Games, and VR for Learning and Hearing Health," by Selvarani Moodley.	***Poster*** [2B]. "Corticothalamic Mechanisms of Human Auditory Selective Attention," by Ana-Maria Gore.
	Poster [1A]. "From Part Task Training to Integrated Simulation: Addressing Clinical Capacity and Assessment Challenges in Audiology Education," by Selvarani Moodley.	***Poster*** [2B]. "Effects of Environmental Auditory Complexity on First- and Higher-Order Mismatch Negativity: A Test of Hierarchical Predictive Processing," by Sahana Murali.

	Poster [1A]. "Perceived Time vs Objective Time in Audiometric Testing: Fast Bayesian vs Hughson-Westlake Methods in Young Musicians," by Ines Ben Aïssa.	***Poster*** [2B]. "Extending Dynamic Causal Modelling with Interlaminar Connectivity for Laminar fMRI," by Alicia Gonsalves.
	Poster [2A]. "Test–Retest Reliability of the Audible Contrast Threshold (ACT) in Young Normal-Hearing Adults," by Kavya Sreekumar.	***Poster*** [2B]. "Autonomic Responses to Affective Sounds in Misophonia and Hyperacusis," by Naser Salas-Husain.
	Poster [2A]. "Effect of Online Music-Based Workshops on Music Perception in Older Adults with Hearing Loss," by Alexis Whittom.	***Poster*** [3B]. "Exploring Auditory Encoding in Misophonia using Frequency Following Response," by Hamssika Sudhakar.
22:30 – 22:40	Mini-break	
22:40 – 23:15	***Poster*** [2A]. "Extended High-Frequency (EHF) Hearing and Speech-in-Noise Difficulty Despite Normal Conventional High-Frequency Hearing and Preserved Word Recognition in Quiet," by Navid Shahnaz.	***Poster*** [3B]. "Auditory Sensory Gating in Misophonia: A Neurophysiological Evaluation," by P. R. Sujeeth.
	Poster [2A]. "Cadenza: Lyric Intelligibility Prediction for Musical Enjoyment," by Rebecca Vos.	***Poster*** [3B]. "Inference of Cochlear Amplifier Gain Using Simulated Distortion-Product Otoacoustic Emission Level Maps," by Vaclav Vencovsky.
	Poster [2A]. "Effects of stimulus degradation and attention on hierarchical auditory processing: Evidence from brainstem and cortical responses," by Srikanth Nayak.	***Poster*** [3B]. "Age-Related Alteration of Neural Correlates of Tone-in-Noise Detection in the Auditory Midbrain," by Dimitri Brunelle.
	Poster [2A]. "Cortical Speech Tracking in Older Adults with Hearing Loss," by Mridula Sharma.	***Poster*** [3B]. "Multi-Response Deconvolution preserves pABR Estimation Quality Despite Increasing Response Overlap," by Pilar Castillo.
	Poster [3A]. "Investigating the Link Between Sleep Quality, Acoustic exposure, and Proxy Measures of Hidden Hearing Loss in Young Adults," by Sreelakshmi Suresh.	***Poster*** [3B]. "Effect of Context on Cortical Speech Tracking in Young Adults with Normal-Hearing," by Fauve Duquette-Laplante.

	Poster [3A]. "Effect of Aging on Forward-Masking Estimates of Medial Olivocochlear Gain Reduction: Computational simulations and Behavioral Data," by David López Ramos.	***Poster*** [3B]. "Test–Retest Reliability and Agreement of Ear-Canal and Mastoid Wave I Auditory Brainstem Responses in Healthy Young Adults," by Blandine Duval.
	Poster [3A]. "Epidemiology and Profiling of Tinnitus in Adolescents," by Merin Benny.	
	Poster [3A]. "Computational Closed-Loop Methods to Compensate for Standard and Hidden Hearing Losses," by Matthias Inghels.	
	Poster [3A]. "Tuning In to the Phantom Sound," by Jack Stevenson.	
	Poster [3A]. "Speech-in-Noise as a Biomarker of Peripheral Auditory Neuropathy in Metabolic Disorders: Proof of Concept in Patients with Uncontrolled Type 2 Diabetes," by Mathieu Schué	
23:15 – 23:20	**First-day conference closure**	
23:20 – 00:00	**Post-Conference Poster Meet & Connect** *Dedicated breakout rooms will be available for all posters presented in Sessions PB1 and PB2, allowing attendees to meet the authors, discuss the work in more detail, and ask questions.*	

Day 2 (Friday 26th June, 2026)

Time UTC+2	Room A	Room B
12:00	**Second-day conference welcome**	
12:00 – 12:45	**Keynote Lecture 3.** *"Characterizing auditory physiologic responses in pediatric populations at risk"*	

	Prof. Linda J. Hood (Vanderbilt University Medical Center, Nashville, TN, USA) *Presented by Dr. Jan-Willem Wasmann*	
12:45 – 12:55	Mini-Break	
12:55 – 14:10	***Session 4.A. Machine Learning & AI for Hearing Science*** *Intelligent algorithms for diagnosis, prediction, and personalisation of hearing interventions.* Chaired by Prof. Dr. Tobias Goehring.	***Session 4.B. Tele-Audiology and Next-Generation Clinical Tools*** *Remote assessment, AI-assisted monitoring, and digital rehabilitation platforms.* Chaired by Dr. Nikki Philpott.
	Featured. "Perceptual Sensitivity is Explained by Optimization for Ecological Hearing Tasks," by Mark Saddler.	***Featured***. "VIA+: A Gamified Digital Ecosystem for the Integrated Phenotyping of Auditory and Cognitive Development," by Frank Betances Reinoso.
	Podium. "Frequency Following Response in Children with School Difficulties or Stroke: Integrating Traditional Statistics and Machine Learning Techniques," by Jheniffer Raimundo.	***Podium***. "A Digital Ecosystem for Scalable Tinnitus Rehabilitation," by Zahi Tubul.
	Podium. "Mapping the Hearing Loss Landscape: A Data-Driven Phenotyping Framework Beyond Pure-Tone Audiometry," by Gerard Encina-Llamas.	***Podium***. "ULIXES: Universal Language-Independent eXploration of Effects of Speech Cues: Pilot Evidence of SNR-Dependent Multisensory Reweighting as a Novel Clinical Index," by Jorge Mejia.
	Podium. "DeepFS4: A Deep Neural Network-Based Sound Coding Strategy for Cochlear Implants," by Xavier Pérez Ruiz.	***Podium***. "Development, Implementation, and Clinical Benefits of a Sound Preference Tool," by Frederic Marmel.
	Podium. "An AI Voice Agent for Speech Communication Assessment," by Joanna Luberadzka.	***Podium***. "Exploration of the Effectiveness of Online Remote Programming for Domestic Cochlear Implant Systems," by Tianyi Liu.
	Podium. "Inferring Cochlear Cell Damage From Auditory Evoked Potentials Using Physics-Informed Machine Learning," by Miguel Temboury.	***Podium***. "Loudness Scaling with Variations in Remote Mobile Hardware: Can we Trust the Results?" by Thomas Schwarz.

14:10 – 15:30	Lunch / Dinner Break	
15:30 – 16:00	**Invited Speaker 3.** “Extracochlear Electric–Acoustic Interaction as the Basis for a Novel Hearing Device for Restoration and Diagnosis” **Prof. Dr.-Ing. Waldo Nogueira** (Hannover Medical School, Hannover, Germany) *Presented by Dr. María Amparo Callejón Leblic*	
16:00 – 16:10	Mini-Break	
16:10 – 17:25	**Session 5.A. Cochlear Implants: Challenges and Prospects** *Advances in signal processing, computational models, neural interfaces, and personalised stimulation strategies.* Chaired by Prof. Tania Hanekom.	**Session 5.B. Hearing-Aid Innovation and Performance Assessment** *Algorithms, real-ear analysis, and adaptive processing in modern devices.* Chaired by Dr. Raul Sanchez Lopez.
	Featured. “Neural Response-Driven Optimization of Cochlear Implant Stimulation Using Differentiable Auditory Models,” by Julie Van Heghe.	***Featured***. “A Large-Scale Database and Predictive Model of Listener-Rated Ease of Speech Understanding in Commercial Hearing Aids,” by Andrew Sabin.
	Podium. “Comparative Study of AutoNRM Intelligent Mapping and Manual Mapping in Nurotron Cochlear Implant Recipients,” by Yipeng Lin.	***Podium***. “Performance Assessment for DNN-based Speech Enhancement Algorithms,” by Stefan Raufer.
	Podium. “Prediction of Speech Outcomes with Cochlear Implants: Limitations and Opportunities for Clinical Practice,” by Marta Campi.	***Podium***. “Technical Evaluation of Scene-Aware Dynamic Compression for Hearing Aids Using a Database of Natural Conversations,” by Moritz Bender.
	Podium. “From Clinic to Reality: Integrating Sound Field Testing and Hearing Quality Measures in Cochlear Implant Users,” by Marta Álvarez-Cendrero.	***Podium***. “Individualized Amplitude Compression for Improved Listening Outcomes at Moderate to High Sound Levels,” by Lukas Jürgensen.
	Podium. “Clinical Validation of the Web-based Platform AUDITO for Cochlear Implant Patients,” by Andreas Markidis.	***Podium***. “Integrating Neural Network Denoisers in Hearing Aids Under Real-Time Constraints,” by Clara Yaiche.

	Podium. "A combined Computational Model of Hair Cell and Auditory Nerve Responses to Acoustic and Electric Stimulation," by Roman Stracke.	***Podium***. "Participant-in-the-Loop Research on Speech Enhancement for Multi-Talker Scenarios in Hearing Aids," by Melika Kianian.
17:25 – 17:45	Break	
17:45 – 18:15	**Invited Speaker 4.** "The Changing Brain: Cross-Modal Neuroplasticity in Hearing loss and its Clinical Implications" **Prof. Anu Sharma** (University of Colorado Boulder, Boulder, CO, USA) *Presented by Dr. Joaquin T. Valderrama*	
18:15 – 18:25	Mini-Break	
18:25 – 19:40	**Session 6.A. From Lab to Clinic: Ensuring Impact of Computational Hearing Research** *Translation, reproducibility, and open-science frameworks for clinical relevance.* Chaired by Dr. Ángel Ramos de Miguel.	**Session 6.B. Multisensory and Vestibular Integration** *Balance, cross-modal perception, and earable technologies.* Chaired by Dr. Seba Ausili.
	Featured. "PECAP Online: Translating the Panoramic ECAP Method to Clinic," by Charlotte Garcia.	***Featured***. "Audio-Visual Speech Recognition: Effects of Auditory Masking and Visual Interference for Healthy Young Adults," by Emma Søndergaard Pedersen.
	Podium. "Reliability and Validity of the Audiometric Weber Test," by Michael Finkelstein.	***Podium***. "Exploring the Auditory-Tactile Integration of Complex Acoustic Features using the Multichannel Vibrotactile Gloves," by Loonan Chauvette.
	Podium. "Toward Ecologically Valid Hearing Assessment," by Zahi Tubul.	***Podium***. "The Effects of Cochlear Implants on the Vestibular System," by Christian Steyn.
	Podium. "Data Standards in Audiology: Data Models and Community Participation," by Mareike Buhl.	***Podium***. "Effects of Chiropractic Interventions on the Audio-Vestibular System: A Systematic Review," by Beyza Yılmaz.

	Podium. "From Detection to Understanding: Speech-in-Noise Screening Identifies Functional Auditory Deficits with Electrophysiologic Correlates in a Clinical Subset," by Jacqueline Scholl.	
19:40 – 20:30	Lunch / Dinner Break	
20:30 – 21:15	**Keynote Lecture 4.** *"Computational auditory models in the service of improving audiology"* **Prof. Enrique A. Lopez-Poveda,** University of Salamanca, Salamanca, Spain *Presented by Dr. Miriam I. Marrufo-Pérez*	
21:15 – 21:25	Mini-Break	
21:25 – 22:00	**Session PB3.** Poster Blitz Presentations from Sessions 4A, 5A and 6A. *Chaired by Dr. María Amparo Callejón Leblic*	**Session PB4.** Poster Blitz Presentations from Sessions 4B, 5B and 6B. *Chaired by Dr. Jan-Willem Wasmann*
	Poster [4A]. "Predicting Early Cochlear Implant Outcomes in Adults: A Machine Learning Approach Using Cognitive and Linguistic Measures," by Benjamin Burns.	***Poster*** [4B]. "Knowledge and Attitudes Toward Tele-Audiology among Hearing Care Professionals," by Hande Çakıroğlu.
	Poster [4A]. "Machine Learning-Based Prediction of High-Frequency Hearing Risk in Airport Catering Personnel Using Demographic, Occupational, and Baseline Audiometric Variables," by Şeyma Öztürk.	***Poster*** [4B]. "Developing RNID's Vision for the Future of Hearing Healthcare," by Lola Russell.
	Poster [4A]. "EdgeConvGRU: A Lightweight Hybrid Architecture for Robust Acoustic Scene Classification for Medical Hearing Aids," by Soumen Sinha.	***Poster*** [4B]. "Comparative Study of Auditory Gestalt Perception Using AI-Generated Clear Speech in Sports and Non-Sports Trainees," by Sneha Chandra Sekar.
	Poster [4A]. "Intelligent System for the Classification and Segmentation of Middle Ear	***Poster*** [5B]. "Do Premium Hearing Aids Offer Better Outcomes Than Basic Models? A Systematic Review

	Pathologies," by Roua Kammoun.	and Meta-Analysis," by Nausheen Dawood.
	Poster [4A]. "The Differentiable Auditory Loop (DAL): Neural Activity Alignment for Personalized Hearing Aids," by Jason Mikiel-Hunter.	***Poster*** [5B]. "Adult Hearing Aid Users' Perspectives on Outcome Expectations and Perceived Value for Money: A Qualitative Study Using the Health Belief Model," by Nausheen Dawood.
	Poster [5A]. "Longitudinal Analysis of Cochlear Implant Stimulation Levels in the First Year: Optimizing Adult Fitting," by Maryam Hussain.	***Poster*** [5B]. "Voice Cues and Vocal Emotions Perception in Adult Hearing Aid Users," by Kateryna Skupovska.
	Poster [5A]. "Optimizing Amplitude Modulation Enhancement for Music Transmission Through Cochlear Implants," by Luuk Schipper.	***Poster*** [6B]. "The Digital Triad: Screen Time, Sleep Disruption, and Vestibulo-Cochlear Dysfunction -A Pilot Study," by Jerlin Glites.
	Poster [5A]. "Towards Modelling the Impact of Spread of Excitation on Classical Psychophysical Paradigms with Cochlear Implant Users," by Rutendo Chatiza.	***Poster*** [6B]. "Multisensory Balance Assessment in Patients with Muscle Tension Dysphonia," by Azra Sivari.
	Poster [6A]. "Data Modelling for openEHR Data Standards in Audiology: Archetypes for three typical Audiological Tests," by Mareike Buhl.	
	Poster [6A]. "Hearing Data: Unlocking the potential of the Noah database," by Nicolas Vannson.	
22:00 – 22:15	**Awards & Closing ceremony**	
22:15 – 23:00	**Post-Conference Poster Meet & Connect** *Dedicated breakout rooms will be available for all posters presented in Sessions PB3 and PB4, allowing attendees to meet the authors, discuss the work in more detail, and ask questions.*	

Keynote speakers

Keynote speaker 1: “Separating the Causes of Listening Difficulties when Thresholds are Normal: Concepts and Data”

Prof. Harvey Dillon, Macquarie University, Sydney (Australia).

Abstract: Listening difficulties in the context of normal thresholds have multiple causes. These include deficits in several different binaural and central auditory processes, speech phoneme identification, language ability, working memory and attention. Determining the dominant cause(s) of the listening difficulty experienced by an individual is complex because individual tests aiming to measure one of these abilities invariably also rely on other abilities. Additionally, multiple deficits can co-occur and some deficits can lead to other deficits.

This talk will outline a non-controversial model of speech understanding in noise, and describe a novel cohesive set of tests built around the model and designed to estimate the degree of deficit in type of ability. Relationships between the abilities measured with these tests will be illustrated with analyses of several data sets, including a very large-scale data set gathered from multiple clinics using these tests. Remediation methods for different types of deficits will also be reviewed.

Biography: Harvey Dillon is currently a part-time Professor at Macquarie University. For the last eight years he has mostly focused on conducting and supervising research into auditory processing disorders and other causes of listening difficulties, including creating new tests for these conditions. He is the author of over 300 scientific publications and a widely used text book on hearing aids. He has been closely associated with various NAL prescription rules, COSI outcomes evaluation, the trainable hearing aid, the LiSN-S test of spatial processing, and clinical cortical response testing. He was previously the Director of the National Acoustic Laboratories, and a part-time Professor at the University of Manchester.

Keynote speaker 2: “Hijacking and helping auditory attention: How interruptions and prediction shape everyday communication”

Prof. Barbara Shinn-Cunningham, Carnegie Mellon University, Pittsburgh, Pennsylvania (USA).

Abstract: Communicating in social settings relies on the ability to focus on a relevant sound amid competing sources—for example, following a dinner conversation at a crowded restaurant. In healthy listeners, volitional, top-down goals and involuntary, bottom-up interruptions interact to prioritize behaviorally important sounds while maintaining situational awareness. In noisy environments, predictive processes can support this effort by helping listeners parse complex auditory scenes. This talk reviews behavioral and neurophysiological research on the brain networks engaged during auditory attention, with a focus on how bottom-up interruptions disrupt top-down focus and how expectation and learning help stabilize it. Together, these findings highlight how top-down and bottom-up attention, memory, and experience interact to support robust communication in everyday environments.

Biography: Barbara is an auditory neuroscientist who combines behavioral, neuroimaging, and computational methods to understand how listeners communicate in everyday environments. She currently serves as the Glen de Vries Dean of Mellon College of Science at Carnegie Mellon University (CMU). A Professor in CMU’s Neuroscience Institute, she also holds courtesy appointments in Psychology, Biomedical Engineering, and Electrical and Computer Engineering. She is a Fellow of the Acoustical Society of America (ASA) and the American Institute for Medical and Biological Engineering. She has received ASA’s Helmholtz-Rayleigh Interdisciplinary Silver Medal, the Brown Engineering Alumni Medal, and research fellowships from the Alfred P. Sloan Foundation, the Whitaker Foundation, and the Vannevar Bush Fellows program. Her mentorship has been recognized by the ASA and the Society for Neuroscience.

Keynote speaker 3: "Characterizing auditory physiologic responses in pediatric populations at risk"

Prof. Linda J. Hood, Vanderbilt University Medical Center, Nashville, TN (USA).

Abstract: Children with prenatal risk factors may be at risk for compromised neural function than can impact development of processes dependent on intact neural function. The primary focus of this presentation will be on results of an ongoing, longitudinal study of auditory physiologic responses from the brainstem to cortex and an extensive battery of language and cognitive measures in children with prenatal Zika virus exposure. A particular focus of this work is on identification of those children without overt characteristics of viral exposure sequalae who remain at potential risk. The value of longitudinal datasets, well-characterized peripheral function, sensitive assays, and a comprehensive battery of measures will be highlighted.

Biography: Linda J. Hood, Ph.D., is a Professor and Hearing Scientist in the Department of Hearing and Speech Sciences and Director of the Auditory Physiology Laboratory at Vanderbilt University, Nashville, Tennessee, USA. Dr. Hood's research career, supported primarily by research grants from the National Institute on Deafness and Other Communication Disorders (NIDCD) focuses on auditory physiology in peripheral and central systems across the lifespan, auditory neuropathy, efferent auditory function, hereditary hearing loss, and comparative hearing studies. Her research is reflected in peer-reviewed scientific publications, textbooks, chapters, invited review articles, and numerous scientific presentations at national and international venues. Dr. Hood participates in review panels of the NIDCD, the World Health Organization, and is an Associate Editor for the journal Ear and Hearing. Dr. Hood is a Past President of the American Academy of Audiology, the American Auditory Society, and the International Society of Audiology. She has served as an Honorary Professor at the University of Queensland, Australia, a visiting professor at the University of Hong Kong, and an International Key Scientist with the Australian Hearing Collaborative Research Centre. Dr. Hood has received the Honors of the Association from the American Speech-Language-Hearing Association, the Jerger Career Award for Research in Audiology from the American Academy of Audiology, the Hallowell Davis Lectureship from the International Evoked Response Audiometry Study Group, the Aram Glorig Award from the International Society of Audiology, and the Life Achievement Award from the American Auditory Society.

Keynote speaker 4: “Computational auditory models in the service of improving audiology”

Prof. Enrique A. Lopez-Poveda, University of Salamanca, Salamanca, Spain

Abstract: Auditory computational models are algorithmic frameworks that simulate how sounds are processed by the auditory nervous system to produce perception. In other words, models are testable working hypotheses about how the anatomy and physiology of the auditory system explain psychoacoustical observations. Models serve to elucidate key physiological processes, acoustic cues, and other factors that determine perception. Beyond their theoretical significance, models may be used as tools to improve audiological diagnosis and treatment. In this presentation, I will review over three decades of research dedicated to the development of auditory models. I will describe our efforts to leverage these models to improve hearing aids and cochlear implants. Finally, I will discuss approaches for developing models that account for listener-specific differences in perceptual outcomes, thereby supporting their successful application in the emerging field of “precision audiology”.

Biography: Enrique A. Lopez-Poveda is a Professor of Audiology and Director of the Auditory Computation and Psychoacoustics Laboratory at the University of Salamanca (USAL). He holds a BSc in Physics (USAL) and a PhD in Hearing Science from Loughborough University (UK). He is the author of three books, three patents, and over 110 publications on hearing sciences. He has been on the editorial board of JASA, JARO, Trends in Hearing, and Ear and Hearing. He has been honored with the Cross of Naval Merit (1999), the Salamanca Convention Bureau Ambassadors Award (2013), the USAL Alumni Award (2013), the Medical Award of the Spanish Federation of Associations of Cochlear Implantees (2022), the María de Maeztu Award for Scientific Excellence from USAL (2023), and the II Enrique Salesa Batlle National Audiology Award (2024). He is a fellow of the Acoustical Society of America, and the International Collegium of Rehabilitative Audiology.

Invited speakers

Invited speaker 1: "A Novel Gamified Neurofeedback System for Enhancing Attention in Naturalistic Listening Environments in Normal-Hearing and Cochlear Implant Users"

A/Prof. Andrew Dimitrijevic, University of Toronto, Toronto (Canada).

Abstract: Listening in noisy natural environments recruits both sensory and cognitive systems including attention and working memory. Individual variations in cognitive systems may underly some of the patient outcome variability observed after cochlear implant surgery. We developed a novel attention-based neurofeedback system designed to reward attentive listening to natural conversations. Using real-time EEG decoders, we show that a videogame-inspired neurofeedback system can be deployed in both normal hearing and cochlear implant users. Our data suggest that such systems can be used in rehabilitation after cochlear surgery and may promote hearing associated neural plasticity and learning.

Biography: Andrew Dimitrijevic is an Associate Professor with affiliations at the University of Toronto and Wayne State University. His research focusses on understanding the neural mechanisms associated with listening in people with hearing loss including those with cochlear implants. He uses high-density EEG to develop neural models to understand listening in complex environments including the effects of attention, working memory and neural plasticity. His research also involves the use of neurofeedback and neuromodulation.

Invited speaker 2: "Hidden Hearing Loss: From Animal Models to Human Translation"

Prof. Sharon Kujawa, Harvard Medical School, Massachusetts Eye and Ear, Boston (USA).

Abstract: Synaptic contacts between sensory inner hair cells and afferent cochlear neurons are highly vulnerable elements in multiple etiologies of acquired sensorineural hearing loss. The cochlear synaptopathy that results has been ubiquitous in all mammals evaluated thus far. It has been labeled "hidden hearing loss", because large synaptic and neural losses can be present in ears with normal or recovered thresholds, creating significant diagnostic challenges. We have characterized noise- and age-related cochlear synaptopathy in ears with isolated neural vs. neural + sensory cell injury/loss. With our collaborators, we have assessed the role of excitotoxicity in producing acute effects of noise and instigating long-term consequences. Our cochlear modeling efforts will aid predictions in human ears where we do not have access to the underlying pathology, and allow tests of our predictions in animal models, where we do. In parallel studies, diagnostic assays in development show great promise in providing clues to cochlear synaptic and neural loss in humans, critical to diagnosis, identification of treatment candidates and assessment of treatment efficacy.

Biography: Sharon G. Kujawa, Ph.D. is a Professor of Otolaryngology-Head and Neck Surgery, Harvard Medical School and a Senior Scientist in the Eaton-Peabody Laboratories, Massachusetts Eye and Ear in Boston, Massachusetts. She serves on the faculty of the Speech and Hearing Biosciences and Technology Program at Harvard University. Prof. Kujawa is a clinician and an auditory neuroscientist whose current work focuses on clarifying how noise and aging cause loss of cochlear synapses and neurons, determining the functional consequences of that loss, and how the degeneration can be manipulated pharmacologically to reveal mechanisms and suggest treatments.

Invited speaker 3: “Extracochlear Electric–Acoustic Interaction as the Basis for a Novel Hearing Device for Restoration and Diagnosis”

Prof. Dr.-Ing. Waldo Nogueira, Hannover Medical School, Hannover (Germany).

Abstract: For severe-to-profound hearing loss, hearing aids are often ineffective, making cochlear implantation (CI) the main treatment. However, CI surgery can damage residual hearing, which strongly enhances speech perception, particularly at low frequencies. We propose a novel approach based on extracochlear electric stimulation (EES), delivering electrical signals via an electrode outside the cochlea to avoid invasive insertion. This enables electric–acoustic stimulation (EAS) by combining low-frequency acoustic amplification with electrical stimulation of non-functional cochlear regions. Importantly, electric–acoustic interactions may provide diagnostic information for assessing low-frequency residual hearing, where current diagnostics are limited. This study evaluates the effectiveness of EES using an active ear-canal electrode or a transtympanic electrode with different reference electrode placements. We assess optimal stimulation configurations and their ability to evoke auditory sensations across frequencies from 250 Hz to 4 kHz. Participants include five normal-hearing adults, five adults with high-frequency hearing loss, and five CI users. Electrical stimuli consisted of 500 ms sinusoids. Perceived loudness and side effects were rated on 0–10 scales, and threshold and maximum comfortable levels were recorded for each condition. Results demonstrate that EES can elicit auditory sensations in all groups, with loudness growth and side-effect profiles depending on frequency and reference electrode placement. Results from the study show that combined EES and acoustic stimulation improved consonant identification compared to acoustic stimulation alone. Moreover, interactions between electric and acoustic stimulation suggest potential for EES-based diagnostic assessment of low-frequency hearing. These findings support the development of non-invasive hearing devices and diagnostics bridging hearing aids and cochlear implants.

Biography: Waldo Nogueira is Professor at Hannover Medical School, where he leads the Auditory Prosthetics Group within the Cluster of Excellence Hearing4all as Principal Investigator. He is also an ICREA researcher hosted by the Institut de Recerca Sant Pau, where he is establishing the Barcelona Hearing Center, and holds a full Professorship at the Universitat Autònoma de Barcelona in the Department of Microelectronics and Electronic Systems and the Institute of Neuroscience. He has received an ERC Consolidator Grant for the READIHEAR project and an ATRAE grant from the Spanish Research Agency for the NEURO HEAR project. His research focuses on technologies for hearing diagnostics and auditory implants, with particular emphasis on the fundamental mechanisms underlying electrical–acoustic interactions in the auditory system. His methodological expertise includes signal processing, psychoacoustics, electrophysiology, and computational modeling. His teaching activities cover signal processing, integrated circuits, hearing technology, and audiology.

Invited speaker 4: “The Changing Brain: Cross-Modal Neuroplasticity in Hearing loss and its Clinical Implications”

Prof. Anu Sharma, University of Colorado Boulder, Boulder, CO (USA).

Abstract: One of the most remarkable aspects of the brain is neuroplasticity, or the brain’s ability to adapt in response to change. Sensory deprivation, as in hearing loss or deafness, results in both structural and functional changes in the brain. One form of compensatory plasticity seen in sensory deprivation is cross-modal neuroplasticity. Cross-modal plasticity is a textbook example of the brain’s ability to reorganize based on use. Cross-modal plasticity refers to the recruitment and repurposing of neuronal resources of a derived sensory modality by an intact modality. In hearing loss, vision and somatosensation recruit and repurpose auditory cortical areas resulting in cortical re-organization. Cross-modal plasticity represents a neuronal process that is dynamic, versatile and reversible and can be exploited for improving clinical outcomes after neurosensory restoration with hearing aids and/or cochlear implants in persons with hearing loss.

Biography: Anu Sharma is Professor and Associate Chair in the Department of Speech Language and Hearing Science, and Fellow in the Institute for Cognitive Science and Center for Neuroscience at University of Colorado, Boulder, USA. Her research focuses on examining cortical neuroplasticity and neurocognitive outcomes in children and adults with hearing loss who are fitted with hearing aids and/or cochlear implants. Her research is funded by the United States National Institutes of Health.

Session 1.A. Computational Audiology & Digital Hearing Research.

Model-based approaches, auditory simulations, and data-driven insights into hearing processes.

Chaired by Dr. Helia Relaño Iborra.

The Effect of Hearing Loss on the Characteristics of Daily Life Conversations

Carlota Sabaté Cao (Amsterdam University Medical Centers)*; Nicole A. Huizinga (Vrije University Amsterdam); Laura Keur-Huizinga (Amsterdam University Medical Center); Jule Pohlhausen (Jade University of Applied Sciences); Eco J.C De Geus (Vrije University Amsterdam); Dorothea Wendt (Eriksholm Research Center, Oticon A/S); Jeppe H. Christensen (Eriksholm Research Center, Oticon A/S); Niek J. Versfeld (Amsterdam University Medical Center); Sophia E. Kramer (Amsterdam University Medical Center); Adriana A. Zekveld (Amsterdam University Medical Center)

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Featured talk

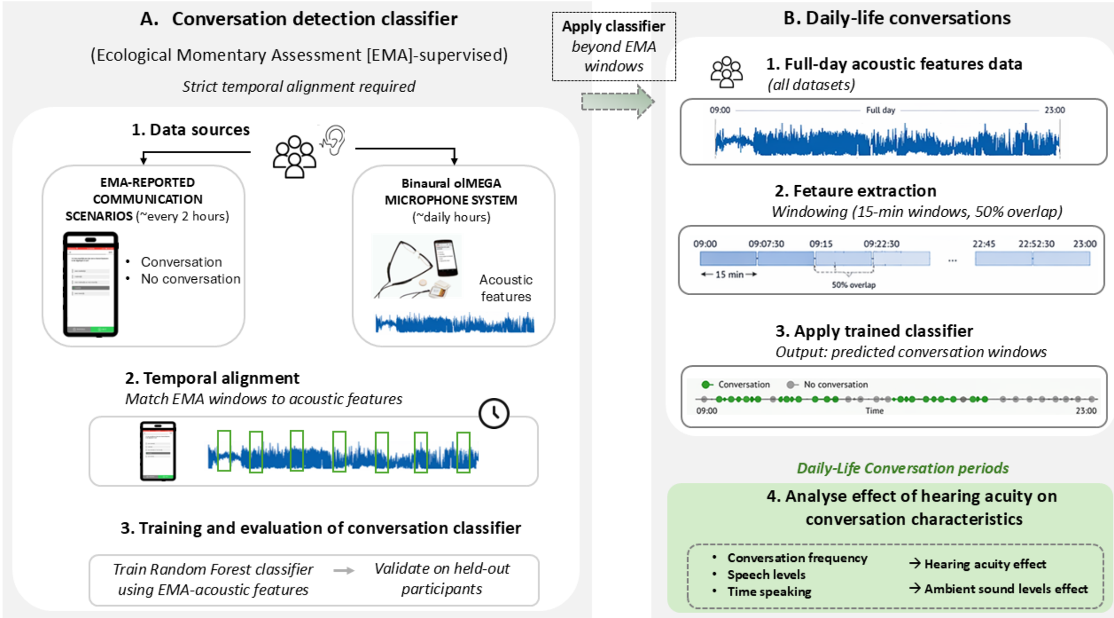
Abstract

Background. Advances in acoustic feature extraction and machine learning enable conversation detection in real-world settings. This provides unique insights into the acoustic characteristics associated with communication. This study assessed the influence of hearing acuity on conversation frequency, own speech levels, and speaking time.

Methods. Acoustic features and Ecological Momentary Assessment (EMA) data were used to detect daily conversations, without accessing raw audio, in 90 participants (aged 40-73 years, mean 59 years) with varying levels of hearing acuity (Pure Tone Average thresholds [PTA] of the better ear, which ranged between -5 and 94 dB HL, mean 28 dB HL). Using the binaural olMEGA microphone system and a random forest classifier trained on EMA-prompted self-reported conversation periods, the likelihood of conversation was predicted throughout the day. The participants' voice was separated from ambient sound and own speech level was assessed. The relationships between PTA and hearing loss (yes/no), and conversation characteristics (frequency, speaking time, speech level) were then assessed.

Results. Using leave-one-subject-out cross-validation, the conversation classifier achieved a mean weighted F1 score of 0.73. Conversation likelihood was associated with noisier environments and hearing acuity. During conversations, speech levels increased with ambient noise. Speaking time was also associated with ambient sound levels, and this effect depended on PTA.

Conclusions. This study shows the potential of combining acoustic and EMA data to assess daily communication. The inability to separate background sound from conversation partners' speech and the short monitoring periods remain a key limitation. Future research should address these and explore individual communication strategies to better understand how hearing acuity affects social interactions.


A. Conversation detection classifier
(Ecological Momentary Assessment [EMA]-supervised)
Strict temporal alignment required
1. Data sources
EMA-REPORTED COMMUNICATION SCENARIOS (~every 2 hours)
Conversation
No conversation
Binaural olMEGA MICROPHONE SYSTEM (~daily hours)
Acoustic features
2. Temporal alignment
Match EMA windows to acoustic features
3. Training and evaluation of conversation classifier
Train Random Forest classifier using EMA-acoustic features
Validate on held-out participants
Apply classifier beyond EMA windows
B. Daily-life conversations
1. Full-day acoustic features data
(all datasets)
09:00
Full day
23:00
2. Fetaure extraction
Windowing (15-min windows, 50% overlap)
09:00
09:07:30
09:15
09:22:30
22:45
22:52:30
23:00
15 min
50% overlap
3. Apply trained classifier
Output: predicted conversation windows
Conversation
No conversation
09:00
Time
23:00
Daily-Life Conversation periods
4. Analyse effect of hearing acuity on conversation characteristics
Conversation frequency
Speech levels
Time speaking
Hearing acuity effect
Ambient sound levels effect

Auditory Models for Hearing-loss Compensation – some Candidates

Lars Bramsløw (Demant A/S)*; Szymon Drgas (Poznan University of Technology); Peter Leer (Eriksholm Research Centre)

labw@eriksholm.com

Podium

Abstract

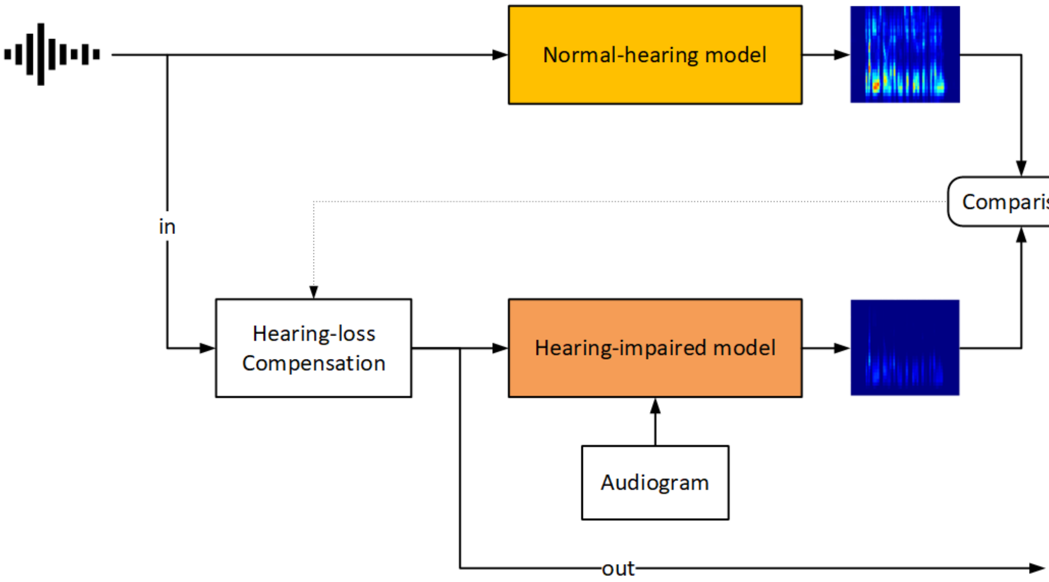
Background. Recent applications of auditory models with hearing loss use 'closed loop' approaches: Deep learning is used to optimize the parameters of a deep neural network (DNN) by comparing a normal hearing and a hearing-impaired branch of the auditory model, thereby obtaining a new, unrestricted (and unknown) hearing-loss compensation (HLC).

The potential of such an approach has already been demonstrated by Leer et al (2025) and Drakopoulos et al (2025), using physiological auditory nerve models. Lately, also psychoacoustic loudness models have been applied for the same purpose (Drgas and Bramsløw, 2026). However, similar model-based approaches have been applied to the development of the NAL-NL1 and NL2 rationales, targeted at conventional multichannel compression hearing aids (Keidser et al, 2011).

Methods. Based on above work and recent other publications, we have made a meta-analysis, comparing some available auditory models, including both quantitative and qualitative outcomes. We present a discussion and some results from these candidate models.

Results. While the physiological models seem to be superior due to the level of detail, they are also computationally demanding and difficult to personalize by audiogram, and supra-threshold measures have yet to be added. Psychoacoustic models are simpler, easier to personalize, but currently also lack important properties of normal and impaired hearing, such as temporal fine structure. For hearing-loss compensation, no model has so far provided clear superior results to another, and only slightly better than conventional NAL-RP linear amplification.

Conclusions. Application of advanced computational auditory models for hearing-loss compensation is still in early days, and there is no obvious preferred model. The existing models are also still lacking important properties of individual hearing losses, so there is more research needed!


Normal-hearing model
Comparison
in
Hearing-loss Compensation
Hearing-impaired model
Audiogram
out

Joint Estimation of Attentional and Masking Error Rates for Speech Recognition in Competing Speech

Adam Bosen (Boys Town National Research Hospital)*

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Podium

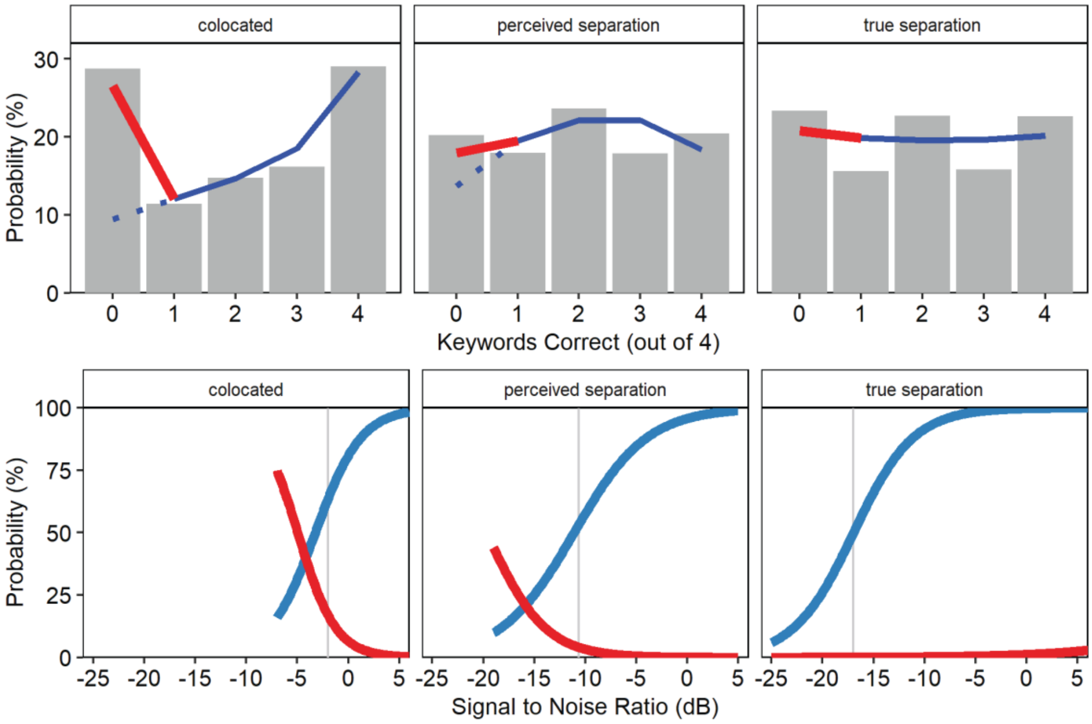
Abstract

Background. Speech recognition in the presence of competing speech is a challenging part of everyday communication. Speech recognition errors in competing speech are often conceptualized as arising from a mixture of energic and informational masking. However, performance on speech recognition tasks is often operationalized as a single outcome measure, such as a percentage of correct responses or a speech reception threshold. The goal of this project is to determine whether analyzing the distribution of recognition accuracy can provide estimates of how often each type of masking occurs.

Methods. Sentence recognition accuracy in competing speech with or without target-masker separation cue (spatial separation and target-masker sex mismatch) of listeners with normal hearing was analyzed using a zero-inflated beta-binomial model. This approach models recognition accuracy as arising from a mixture of two processes. If the listener attends the target then response accuracy is beta-binomially distributed, whereas failures to attend the target add an excess rate of getting zero keywords correct. Mixed effects models of the beta-binomial mean and excess zero rate as a function of signal-to-noise ratio were fit to the data, with random participant intercepts for both model parameters.

Results. The excess zero rate was ~17 % at the average speech-reception threshold when target-masker separation cues were absent, but near-zero when separation cues were present. When separation cues were absent, excess zero rates varied between 5% to 30% across listeners. Random participant intercepts for excess zero rate and beta-binomial mean were uncorrelated and together determined individual speech reception thresholds.

Conclusions. Individuals vary in their susceptibility to informational and energetic masking when recognizing speech in the presence of competing speech. These distinct aspects of performance can be jointly estimated with zero-inflated beta-binomial models.



Listeners heard sentences that contained 4 keywords in the presence of a two-talker same-sex masker that was either colocated with the target, perceptually separated by the precedence effect, or physically separated from the target at +90 degrees azimuth across a range of signal-to-noise ratios. The top row shows model response distribution at the median signal-to-noise ratio when participants attended the target in blue, with zero-inflation adding additional responses at 0 of 4 keywords correct in red. For comparison, the proportion of number of correct responses within 2 dB of the average 50% speech reception threshold is shown in gray bars. The bottom row shows how excess zero rate (in red) and mean accuracy when attending the target (in blue) vary as a function of signal-to-noise ratio. The horizontal gray line shows the average speech reception threshold for each condition.

Measurement-Aware Registry Design for Cochlear Implant Outcomes: Lessons from a 25-Year Clinical Dataset

Apurv Shukla (University of Michigan)*; Amy Paolleti (University of Michigan Medicine); Pravansu Mohanty (University of Michigan, Dearborn); Devin McCaslin (University of Michigan Medicine)

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Podium

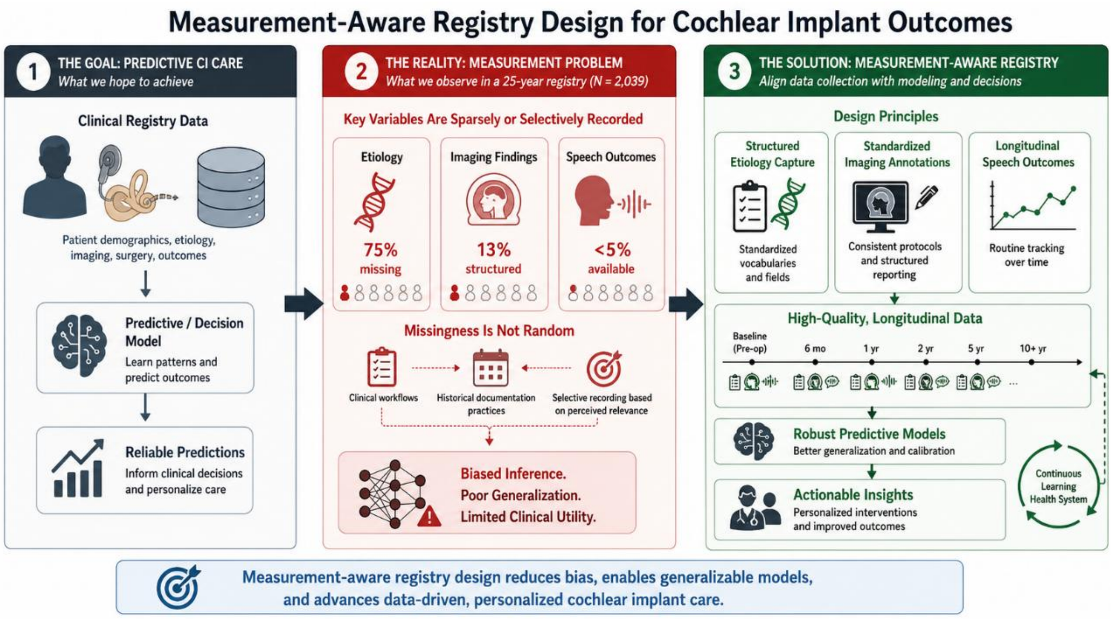
Abstract

Advances in computational audiology increasingly rely on real-world clinical data to develop predictive and decision-support models for cochlear implant (CI) care. However, the structure and completeness of existing clinical registries remain poorly aligned with these goals. In particular, key variables—such as etiology, imaging findings, and longitudinal speech outcomes—are often inconsistently documented, introducing systematic measurement bias that limits downstream modeling.

We analyze a 25-year single-center cochlear implant registry comprising 2,039 adult patients to characterize the data-generating and measurement processes underlying CI records. While clinically meaningful structure is evident—etiology stratifies patients by age, onset type, imaging findings, and adverse outcomes—we find that critical variables are sparsely or selectively recorded. Etiology is undocumented in 75% of patients, imaging findings are structured in only 13%, and speech outcomes are available for fewer than 5%, despite their central role in evaluating CI success .

These missingness patterns are not random: they reflect clinical workflows, historical documentation practices, and selective recording based on perceived relevance. As a result, naïve analyses risk biased inference, and standard machine learning pipelines are unlikely to generalize.

We argue for a measurement-aware registry design paradigm in computational audiology, where data collection is explicitly aligned with modeling and decision-making objectives. Specifically, we highlight the need for (i) structured etiology capture, (ii) standardized imaging annotations, and (iii) longitudinal speech outcome tracking. Our findings provide both an empirical characterization of current limitations and a concrete roadmap for designing CI registries that support predictive modeling and personalized intervention.



Measurement-Aware Registry Design for Cochlear Implant Outcomes
1
THE GOAL: PREDICTIVE CI CARE
What we hope to achieve
Clinical Registry Data
Patient demographics, etiology, imaging, surgery, outcomes
Predictive / Decision Model
Learn patterns and predict outcomes
Reliable Predictions
Inform clinical decisions and personalize care
2
THE REALITY: MEASUREMENT PROBLEM
What we observe in a 25-year registry (N = 2,039)
Key Variables Are Sparsely or Selectively Recorded
Etiology
75%
missing
Imaging Findings
13%
structured
Speech Outcomes
<5%
available
Missingness Is Not Random
Clinical workflows
Historical documentation practices
Selective recording based on perceived relevance
Biased Inference.
Poor Generalization.
Limited Clinical Utility.
3
THE SOLUTION: MEASUREMENT-AWARE REGISTRY
Align data collection with modeling and decisions
Design Principles
Structured Etiology Capture
Standardized vocabularies and fields
Standardized Imaging Annotations
Consistent protocols and structured reporting
Longitudinal Speech Outcomes
Routine tracking over time
High-Quality, Longitudinal Data
Baseline (Pre-op)
6 mo
1 yr
2 yr
5 yr
10+ yr
Robust Predictive Models
Better generalization and calibration
Actionable Insights
Personalized interventions and improved outcomes
Continuous Learning Health System
Measurement-aware registry design reduces bias, enables generalizable models, and advances data-driven, personalized cochlear implant care.

Software-Assisted Analysis and Classification of Cortical Auditory Evoked Potentials

Manuel Lazo-Maestre (HUVM)*; MªAmparo Callejon-Leblic (HUVM-US)

manu11235@gmail.com

Podium

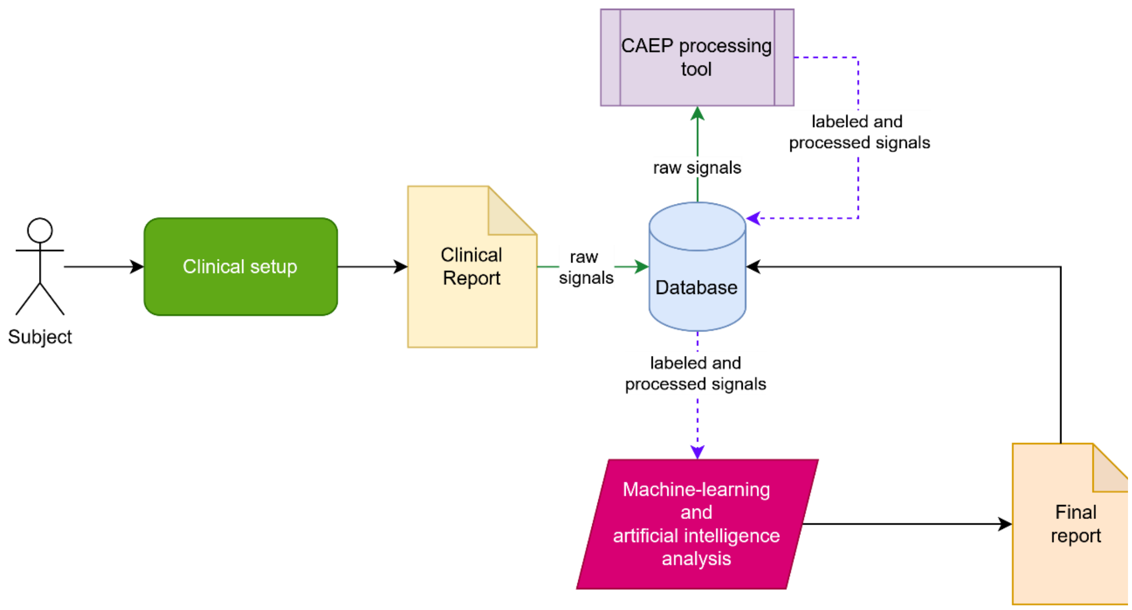
Abstract

This work presents the development and evaluation of a MATLAB-based software system for assisted analysis of cortical auditory evoked potentials recorded using clinical equipment. The tool enables signal labeling based on response presence (clear, inconclusive, or absent) through identification of key waveform components P1, N1, and P2. From these points, relevant features such as latencies, amplitudes, signal-to-noise ratio (SNR), and correlation coefficient (CC) are extracted, supporting both classification and physiological interpretation. The system also manages a structured database of expert-labeled signals.

A total of 742 recordings were used to evaluate 33 machine learning algorithms, including support vector machines (SVM), ensemble methods, neural networks, Bayesian classifiers, and K-nearest neighbors. The dataset was split into training (80%), validation (10%), and testing (10%) subsets. Initial results showed accuracies above 80%, with SVM and ensemble models performing best. However, a bias toward the "clear response" class was observed, with limited ability to identify inconclusive cases, likely due to dataset imbalance and the greater homogeneity of clear signals.

Confusion matrices and ROC analysis confirmed this trend. Partial dependence analysis highlighted the importance of N1-P2 and N1 amplitudes. To improve performance, additional features (SNR and CC) were incorporated, increasing accuracy, with ensemble Bagged Trees models reaching up to 89% on the test set.

These results show that adding features and increasing dataset size can enhance classification, especially for ambiguous cases, supporting automated clinical analysis and improving understanding of cortical auditory responses.


CAEP processing tool
labeled and processed signals
raw signals
Subject
Clinical setup
Clinical Report
raw signals
Database
labeled and processed signals
Machine-learning and artificial intelligence analysis
Final report

Listener-Aware Speech Representations for Hearing-Aid Applications

Sarthak Mangla (Purdue University)*; Afagh Farhadi (Purdue University); Michael Heinz (Purdue University)

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Podium

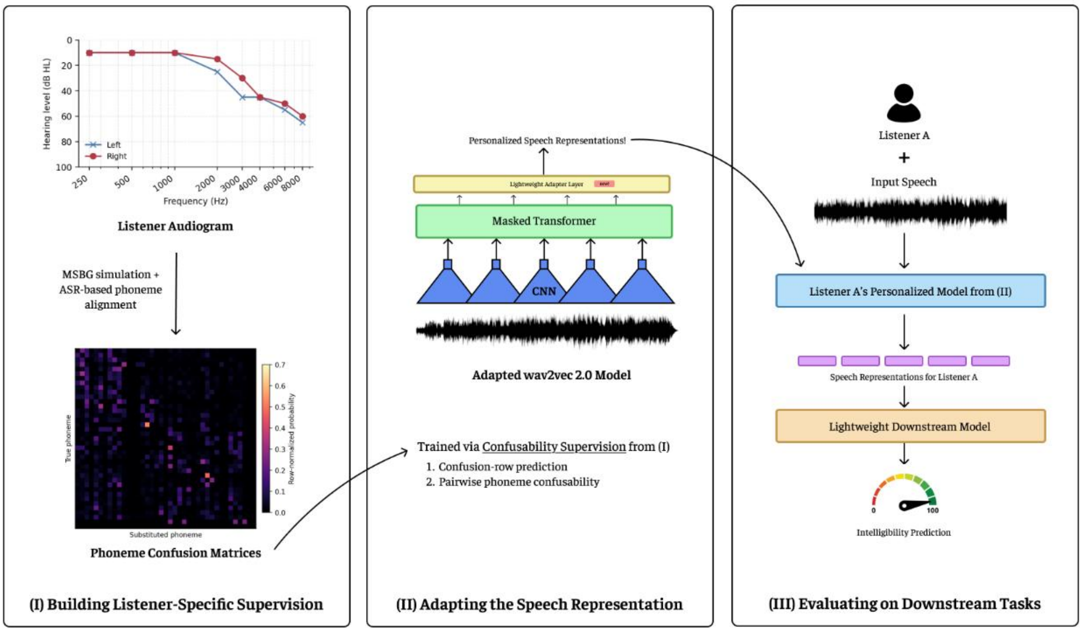
Abstract

Background. Self-supervised speech representations are widely used in hearing-aid enhancement and intelligibility prediction, but are usually learned as listener-independent acoustic features. This limits personalization, since hearing-aid benefit depends on individual hearing profiles and speech-perception patterns. We test whether these representations can be made listener-aware using audiogram-derived phoneme confusability, without requiring behavioral responses.

Methods. We studied 31 Clarity Prediction Challenge listeners with binaural audiograms. For each listener, we synthesized isolated words, degraded them with MSBG simulation and noise, and transcribed them with wav2vec 2.0 ASR. Clean and degraded outputs were converted to TIMIT-39 phoneme sequences and aligned to estimate listener-specific substitutions, yielding row-normalized 39 × 39 confusion matrices. These matrices adapted frozen wav2vec 2.0 embeddings through auxiliary tasks: predicting confusion patterns, estimating pairwise phoneme confusability, or combining both. We evaluated adapted representations on a custom listener-aware benchmark using hearing-aid-processed speech.

Results. Generic self-supervised adaptation on TIMIT did not improve over frozen wav2vec 2.0, while listener-aware objectives improved prediction relative to SSL-TIMIT. The phonetic objective performed best, reducing RMSE from 27.38 to 26.80 and increasing Pearson correlation from 0.674 to 0.700. In paired tests on matched predictions, all listener-aware objectives significantly reduced RMSE compared with SSL-TIMIT: phonetic ΔRMSE = -0.57, $p = 0.004$; pairwise ΔRMSE = -0.39, $p = 0.014$; and combined ΔRMSE = -0.56, $p < 0.001$.

Conclusions. Audiogram-derived phoneme confusability can supervise listener-aware speech representation learning. The strongest gains came from confusion-matrix row prediction, suggesting that hearing-aid personalization can begin at the representation layer using simulated perceptual structure.



Hearing level (dB HL)
Frequency (Hz)
Left
Right
Listener Audiogram
MSBG simulation +
ASR-based phoneme
alignment
True phoneme
Substituted phoneme
Row-normalized probability
Phoneme Confusion Matrices
(I) Building Listener-Specific Supervision
Personalized Speech Representations!
Lightweight Adapter Layer
Masked Transformer
CNN
Adapted wav2vec 2.0 Model
Trained via Confusability Supervision from (I)
1. Confusion-row prediction
2. Pairwise phoneme confusability
(II) Adapting the Speech Representation
Listener A
+
Input Speech
Listener A's Personalized Model from (II)
Speech Representations for Listener A
Lightweight Downstream Model
0
100
Intelligibility Prediction
(III) Evaluating on Downstream Tasks

R-based Systematization of Advanced Audiometric Formulas: Beyond the 25 dB Threshold

Ramón Hernández-Villoria (Centro Clinico de Audición y Lenguaje Cealca)*

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Poster

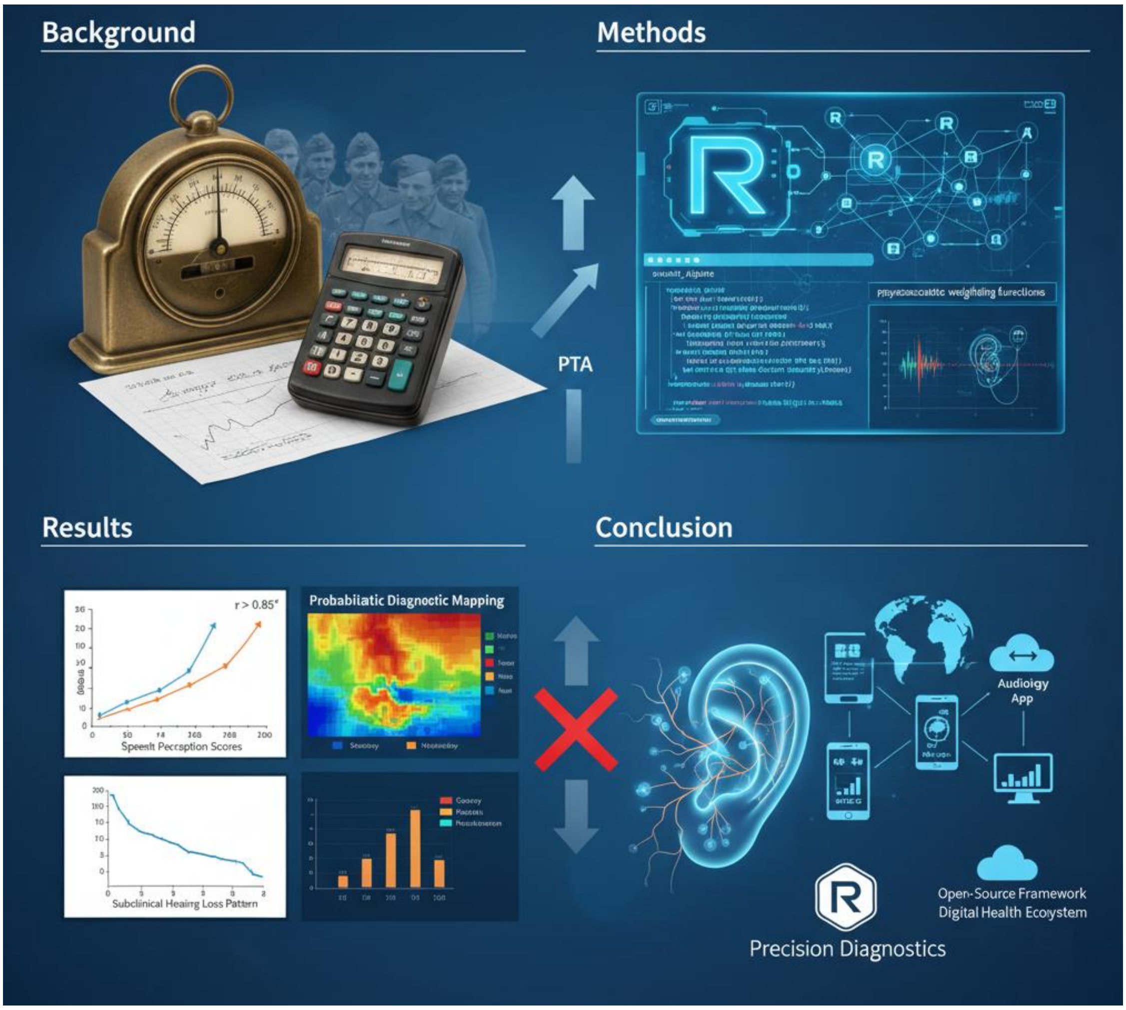
Abstract

Background. The calculations used to obtain a person's hearing level through behavioral tonal audiometry are based on a simple arithmetic average of the threshold values at four frequencies (PTA-4). While this method served to resolve compensation claims for war veterans eighty years ago, even its creators were aware of its limitations. Today, these averages, with 25 decibels considered the upper limit of normal hearing, are still used in most parts of the world.

Methods. The transitions from linear arithmetic to multi-parametric algorithms were implemented in the R environment. We integrate weighted frequency variables, adjusting for age-correction and binaural summation effects. The model incorporates nonlinear regression and psychoacoustic weighting functions that account for spectral smearing and speech-in-noise interference. This R-based framework allows for the processing of raw threshold data into high-resolution functional profiles, moving beyond the static 25 dB cutoff.

Results. The implementation yields specific diagnostic ratios and spectral slope coefficients that demonstrate a higher correlation with speech perception deficits than the traditional PTA-4. Our results indicate that these parameters can identify subclinical hearing loss patterns and provide probabilistic diagnostic mappings. Specifically, the R-systematized ratios facilitate the differentiation between sensory and neural involvement through automated pattern recognition, offering a sensitivity increase in predicting functional disability compared to legacy averaging methods.

Conclusions. The integration of these advanced computational models marks a definitive departure from the "pencil-and-paper" paradigm. By systematizing in R, we provide a scalable, open-access framework. This approach not only empowers manufacturers to develop the next generation of "smart" audiometers but also democratizes precision diagnostics for clinicians worldwide.


Background
Methods
PTA
Results
Conclusion
Probabilistic Diagnostic Mapping
Subclinical Hearing Loss Pattern
Audiology App
Open-Source Framework
Digital Health Ecosystem
Precision Diagnostics

Audiometric data reproducibility with differential privacy using the Laplace mechanism

Hector Gabriel Corrale de Matos (Department of Speech Language Pathology and Audiology of the University São Paulo Campus Bauru)*; Jan-Willem Wasmann (Radboud University Medical Centre); Kátia de Freitas Alvarenga (Department of Speech Language Pathology and Audiology of the University São Paulo Campus Bauru); Lilian Cássia Bórnia Jacob (Department of Speech Language Pathology and Audiology of the University São Paulo Campus Bauru)

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Poster

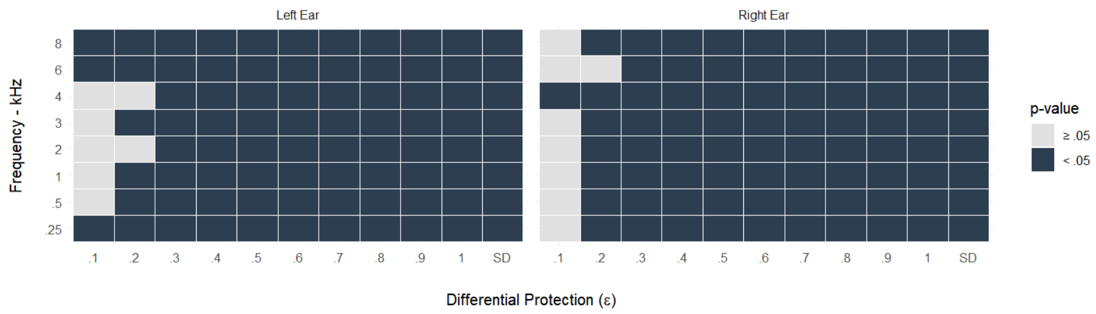
Abstract

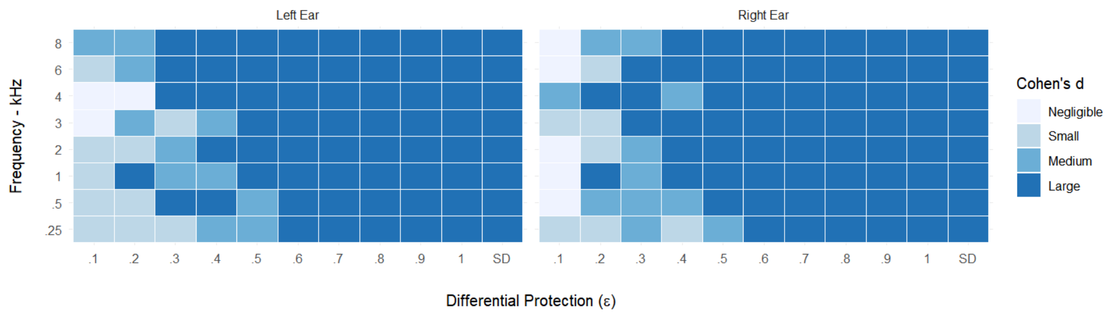
Background. Computational reproducibility relies on open data access. Ethical and legal restrictions on patient privacy often hinder the sharing of raw audiometric data. Differential privacy offers a mathematical framework to share aggregate statistical patterns while guaranteeing individual anonymity. Its use in clinical audiology data utility remains underexplored.

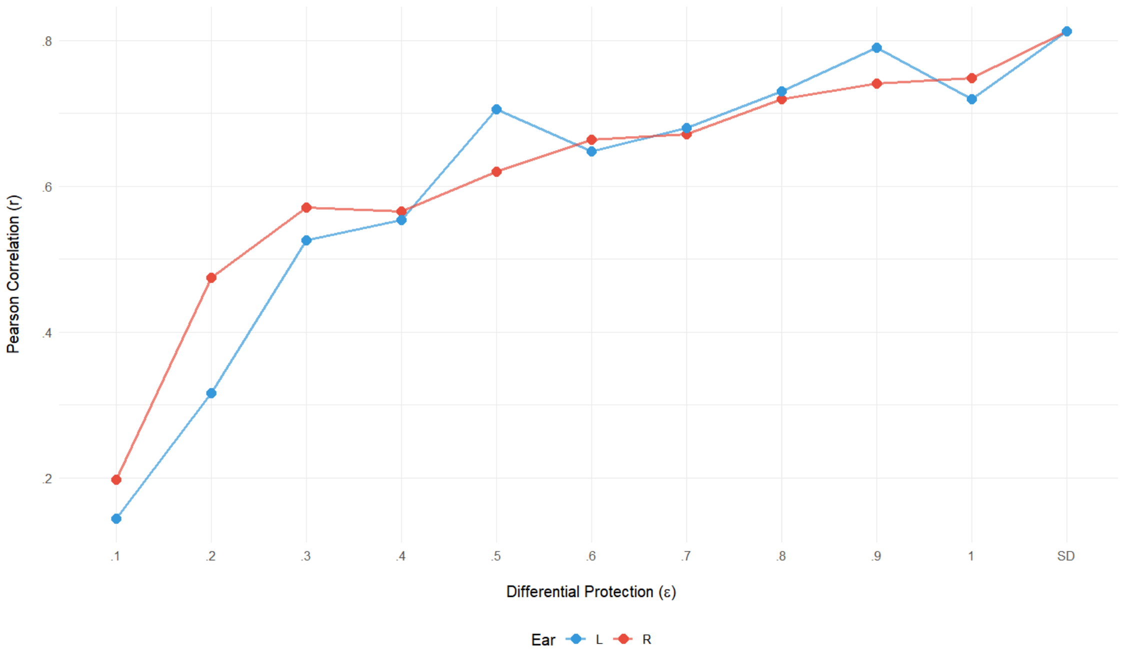
Methods. We generated a synthetic dataset (N=100) simulating a clinical population with two distinct clusters: young adults with normal hearing and older adults with hearing loss. The Laplace mechanism (noise $\sim \mathrm{Lap}(\Delta/\epsilon)$) was applied to mask the simulated variables (age and air thresholds from 250 to 8000 Hz) across a range of privacy budgets (ϵ) [0.1 (more noise) to 1 (less noise)]. Utility loss was compared from the differentially private data against the synthetic baseline sample. We simulated a statistical analysis of data distribution (Shapiro-Wilk test), group differences (Student's t-test), effect sizes (Cohen's d), statistical power, and Pearson correlation (r) between age and the quadritonal mean. Analysis was performed in R (4.4.0) using Quarto.

Results. Laplacian noise significantly altered the data distribution to a non-linear trajectory. Parametric assumption violations are particularly likely when $\epsilon < 0.5$. These noise levels resulted in negligible effect sizes while reducing the age-hearing correlation ($r < 0.4$). Statistical utility was equilibrated with $\epsilon = 0.8$ (elbow curve). Using this ϵ the correlations ($r > 0.72$; baseline: $r = .81$) and effect sizes ($d > 0.8$) were maintained. High post hoc statistical power (>90%) was observed, and significant group differences remained similar to baseline ($p < 0.05$).

Conclusions. Differential privacy is a viable model for audiometric reproducibility. Selecting a balanced privacy budget allows the sharing of reproducible datasets that preserve clinical validity without compromising patient privacy.


Left Ear
Right Ear
Frequency - kHz
8 6 4 3 2 1 .5 .25
.1 .2 .3 .4 .5 .6 .7 .8 .9 1 SD
Differential Protection (ε)
p-value
≥ .05
< .05


Left Ear
Right Ear
Frequency - kHz
8 6 4 3 2 1 .5 .25
.1 .2 .3 .4 .5 .6 .7 .8 .9 1 SD
Differential Protection (ε)
Cohen's d
Negligible
Small
Medium
Large


Pearson Correlation (r)
.8 .6 .4 .2
.1 .2 .3 .4 .5 .6 .7 .8 .9 1 SD
Differential Protection (ε)
Ear L R

Re-examining Auditory Filter Parameters: A Characteristics-Based Framework with Updated Estimates for Humans

Samiya Alkhairy (MIT, KACST)*

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Poster

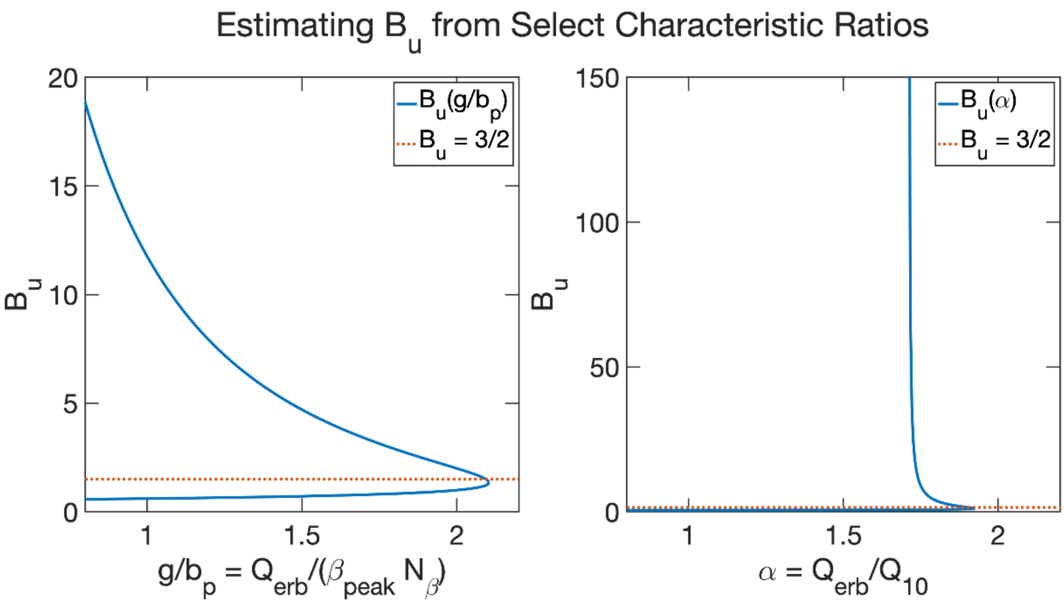
Abstract

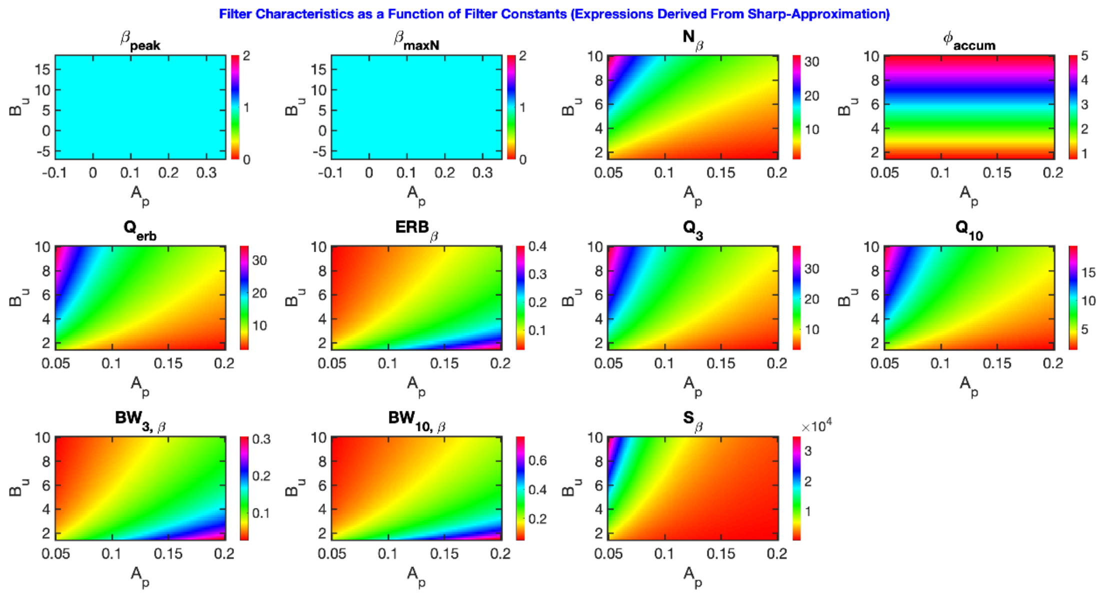
Background. Gammatone-family filters are widely used in computational auditory models, speech technology, and hearing device design. The filter parameters used to mimic human hearing have remained unchanged since Glasberg and Moore (1990), based on simultaneous masking data influenced by suppression. Whether recent forward masking and physiological data warrant updated values has not been systematically investigated.

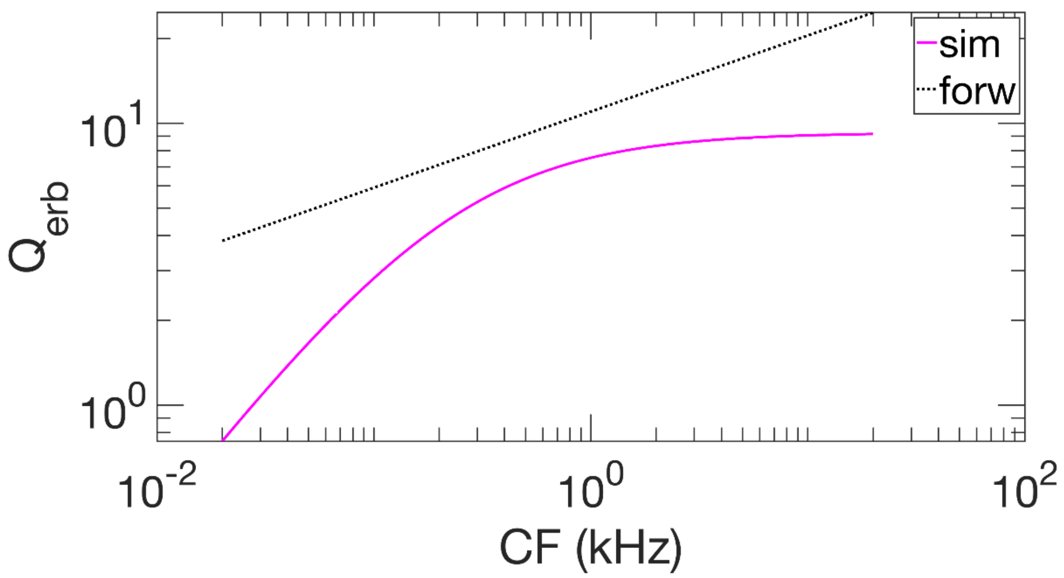
Methods. We developed a characteristics-based framework using a sharp-filter approximation valid across three filter classes of the Gammatone family of filters. We derived closed-form expressions relating measurable characteristics (quality factors, peak group delay, phase accumulation, and spectral convexity) to the filter parameters - pole (Ap), peak frequency ratio (bp), and exponent (Bu). We identified which characteristic ratios are most informative for estimation and applied the framework to recent physiological and psychoacoustic data.

Results. The ratio g=Qerb/Nbeta depends only on Bu and is sensitive enough to yield a point estimate, unlike alpha=Qerb/Q10 which varies too slowly. Using g=1.25 from chinchilla auditory nerve fiber data (Shera et al., 2010), we obtained Bu=7.2, substantially higher than the historical value of 4. Combined with human forward masking quality factors, this yields peak group delays significantly larger than historical estimates at 4 kHz. Relative errors were below 2% across all three filter classes, and the full filterbank requires only 2-3 scalar constants.

Conclusions. The updated parameters have significant implications for predicted temporal envelope resolution, binaural ITD extraction windows, and cochlear implant channel bandwidth-temporal resolution trade-offs. The framework may be extended to serve as the gammatone base of level-dependent models such as the gammachirp and enables systematic parameter updates as new data become available.


Estimating B u from Select Characteristic Ratios
B u (g/b p)
B u = 3/2
B u
g/b p = Q erb /(β peak N β)
B u (α)
B u = 3/2
B u
α = Q erb /Q 10


Filter Characteristics as a Function of Filter Constants (Expressions Derived From Sharp-Approximation)
β peak
β maxN
N β
φ accum
Q erb
ERB β
Q 3
Q 10
BW 3, β
BW 10, β
S β
×10^4
B u
A p


sim
forw
Q erb
CF (kHz)

Capturing Real-World Listening Experiences with Smartwatch-Based Ecological Momentary Assessment

Nicky Chong-White (National Acoustic Laboratories)*; Jhanvi Ramani (National Acoustic Laboratories); Marisa Poulos (National Acoustic Laboratories); Catherine Kwok (National Acoustic Laboratories)

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Poster

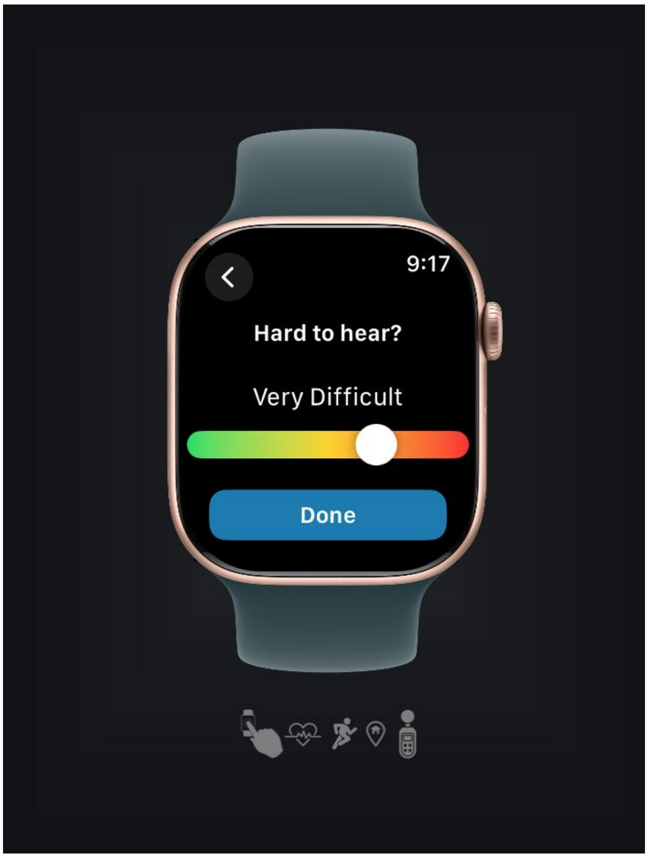
Abstract

Background. Ecological momentary assessment (EMA) is a valuable method for capturing real-world hearing experiences and has evolved from paper-based diaries to smartphone-based approaches. While smartphone EMA enables in-the-moment reporting, it can still be awkward or disruptive in everyday listening situations. Smartwatches offer a more discreet, low-burden way for individuals to report listening difficulties as they occur, while also enabling capture of contextual and physiological data.

Methods. We developed NEMA+ Watch, a smartwatch-based extension of NAL's ecological momentary assessment platform. Participants can initiate brief surveys directly on the Apple Watch when they experience listening difficulty, providing contextual information about the situation. The system also supports collection of passive data streams, including activity, heart rate, and environmental sound exposure, which can be time-aligned with survey responses.

Results. NEMA+ Watch enabled frequent, quick, discreet reporting in everyday environments and supported capture of contextual and physiological data alongside self-reported listening experiences. The platform was successfully deployed in a sponsored research study, demonstrating feasibility and immediate research application.

Conclusions. NEMA+ Watch extends EMA for hearing research by combining user-driven reporting with wearable-based passive data collection. This approach enhances ecological validity and enables richer characterisation of real-world hearing experiences. The platform establishes new capability for multimodal, real-world data capture and supports future research investigating everyday communication challenges.


9:17
Hard to hear?
Very Difficult
Done

A Neural Mass Model of the Auditory Corticothalamic Circuitry

Matthijs Kusters (Dept. of Cognitive Neuroscience, FPN, Maastricht University); Spencer Olson (MaCSBio, FSE, Maastricht University & ExpORL, Dept. Neuroscience, KU Leuven)*; Mario Senden (Dept. of Cognitive Neuroscience, FPN, Maastricht University); Michelle Moerel (MaCSBio, FSE, Maastricht University & Dept. of Cognitive Neuroscience, FPN, Maastricht University)

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Poster

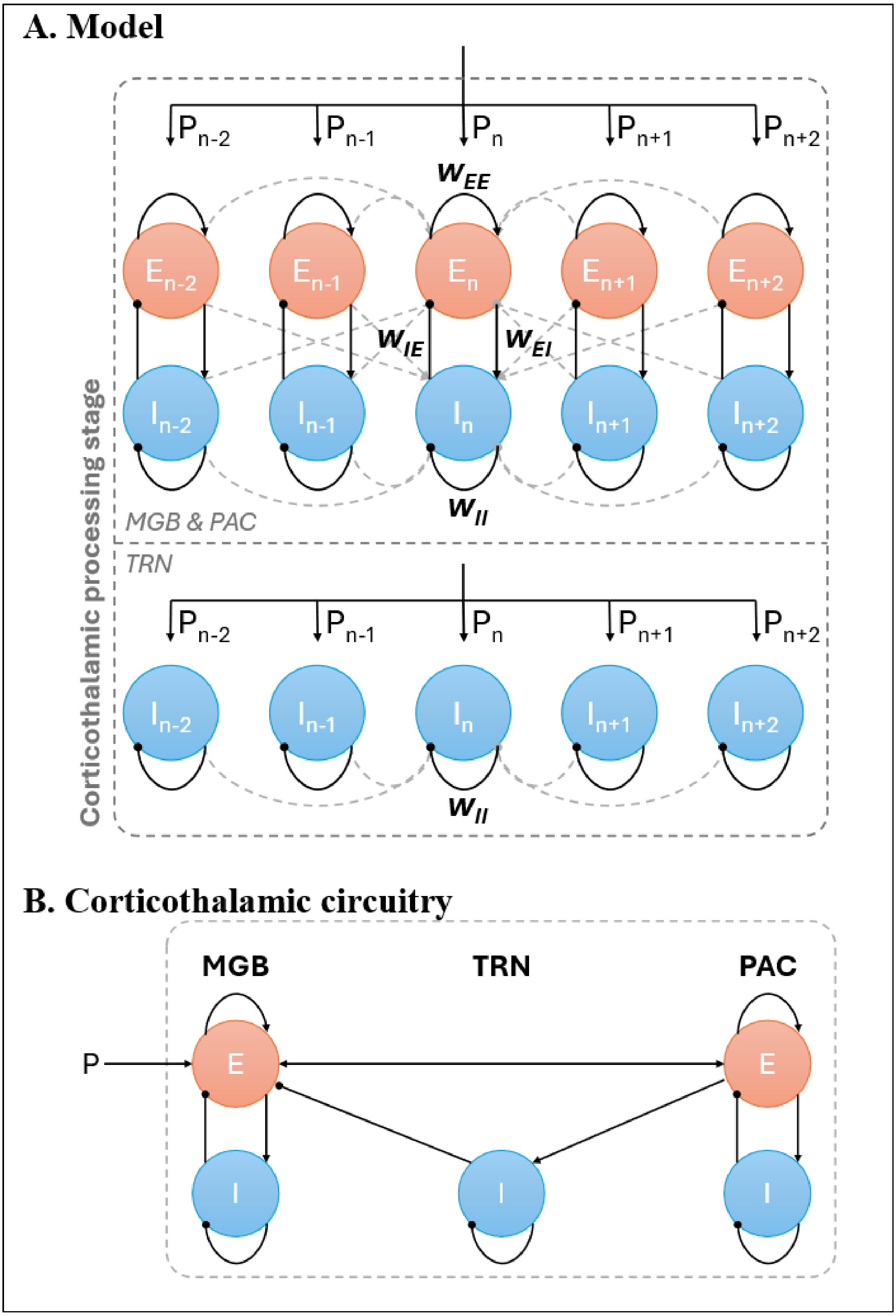
Abstract

Background. Human hearing is remarkably flexible. In crowded environments, we effortlessly attend to relevant sounds while ignoring irrelevant ones. The circuitry between the auditory thalamus (MGB), primary auditory cortex (PAC), and thalamic reticular nucleus (TRN) has been proposed as key for attentive sound processing. Whether this circuitry can support the computations underlying selective attention remains unknown.

Methods. The Wilson-Cowan Cortical Model was adapted to include MGB-PAC-TRN circuitry, with each region represented by inhibitory-excitatory neuron populations (except the TRN, which contains only inhibitory populations). Optimization proceeded in three steps: (1) a sensitivity analysis across 40,000 simulations; (2) model selection based on biological plausibility of firing rates; and (3) parameter tuning to match electrophysiological literature. Robustness was evaluated by monitoring parameter changes on firing rates and tuning.

Results. From 40,000 simulations, 2,710 viable parameter sets were identified and narrowed to a single optimal configuration. The final model produced firing rates and tuning widths consistent with electrophysiological literature, with MGB and PAC exhibiting sharp frequency tuning and TRN showing broader tuning approximately 2.5 times that of MGB. Multiple parameter configurations yielded plausible responses, indicating the model is not constrained to a unique solution.

Conclusions. The implemented MGB-PAC-TRN circuitry demonstrates that the proposed corticothalamic network can be incorporated into a computationally stable model with biologically plausible dynamics. Future work should implement attentional modulation to assess whether this circuitry reproduces neural dynamics observed during selective attention tasks. The model also holds potential for tinnitus research, where TRN dysfunction is hypothesized to allow aberrant signals to propagate into the auditory cortex.


A. Model
Corticothalamic processing stage
P_{n-2}
P_{n-1}
P_n
P_{n+1}
P_{n+2}
W_{EE}
E_{n-2}
E_{n-1}
E_n
E_{n+1}
E_{n+2}
W_{IE}
W_{EI}
I_{n-2}
I_{n-1}
I_n
I_{n+1}
I_{n+2}
W_{II}
MGB & PAC
TRN
P_{n-2}
P_{n-1}
P_n
P_{n+1}
P_{n+2}
I_{n-2}
I_{n-1}
I_n
I_{n+1}
I_{n+2}
W_{II}
B. Corticothalamic circuitry
MGB
TRN
PAC
P
E
E
I
I
I

Student-Led Digital Innovation in Audiology: Virtual Platforms, Games, and VR for Learning and Hearing Health

Selvarani Moodley (University of the Witwatersrand)*; Dhanashree Pillay (University of the Witwatersrand)

selvarani.moodley@wits.ac.za

Poster

Abstract

The rapid expansion of digital technologies presents new opportunities to transform audiology education, clinical training, and public hearing health engagement. This paper presents a suite of student-led digital arts projects developed in collaboration between the Audiology and Digital Arts Departments at Witwatersrand University . The projects focus on:

1) exploring serious games developed for hearing screening and familiarisation with audiological testing. These games target both students and lay users, using play-based learning to demystify hearing assessments, increase user engagement, and promote accurate screening. By integrating core audiological principles into gameplay, the tools support learning while maintaining clinical relevance.

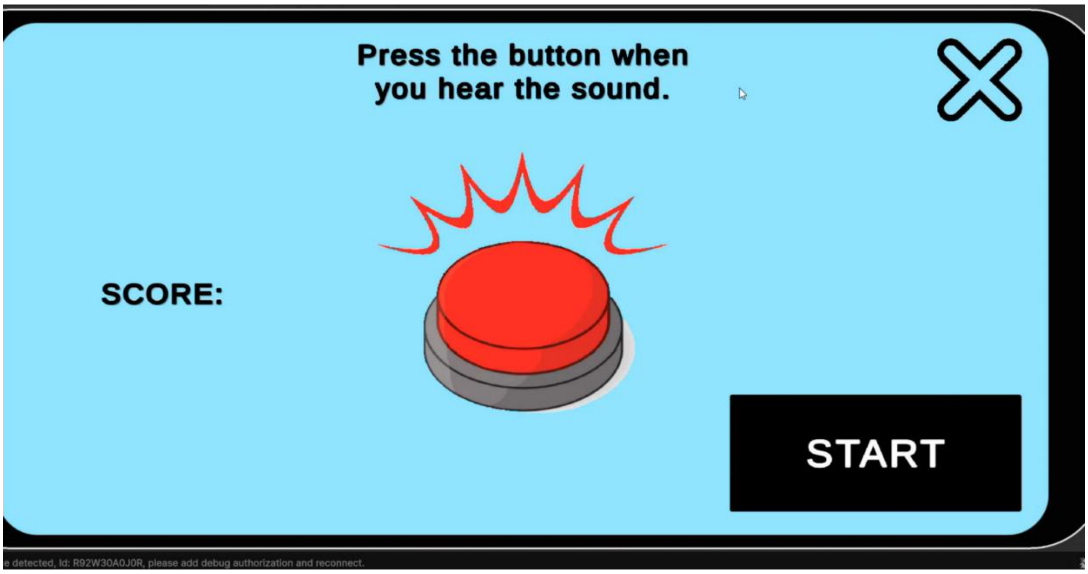
2) a game designed to teach hearing health knowledge alongside introductory sign language. This project addresses both prevention and inclusion, promoting awareness of hearing conservation while fostering communication skills with Deaf and hard-of-hearing individuals.

3) development of a website to showcase hearing-related information and encourage youth to be more sensitive to noise exposure

4) development of a virtual reality tool for vestibular testing. The tool allows the client to explore symptoms while engaged in (virtual reality) scenarios common in vestibular testing and case history questioning.

5) development of a virtual learning platform designed to support students' study skills and early clinical practice. The aim is to design a serious game prototype using established game-based learning principles, and to critically evaluate its alignment with the study of audiology and hearing healthcare, specifically within a South African context.

These projects highlight the value of integrating digital arts, game design, and virtual reality into audiology education. They illustrate how creative digital innovation can enrich learning, support clinical competence, and advance hearing health education in virtual spaces.


Press the button when
you hear the sound.
SCORE:
START
e detected, Id: R92W30A0J0R, please add debug authorization and reconnect.

From Part Task Training to Integrated Simulation: Addressing Clinical Capacity and Assessment Challenges in Audiology Education

Selvarani Moodley (University of the Witwatersrand)*; Dhanashree Pillay (University of the Witwatersrand)

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Poster

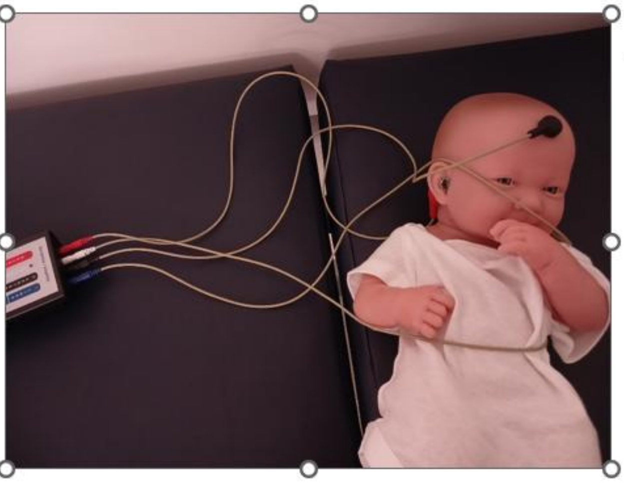
Abstract

A significant challenge in the training of medical students is the increasing student numbers requiring clinical exposure, coupled with the limited capacity of academic hospitals to provide adequate training opportunities. This challenge is equally applicable to the education of audiology students and other allied health professionals.

Alongside limitations in clinical training capacity, audiology education faces challenges in assessing student competence, particularly in clinical areas that rely on less clearly defined criteria and include non technical skills such as case history acquisition, clinical reasoning, and patient feedback. While quality healthcare education requires integrated and holistic skill development, effective competency acquisition necessitates the intentional deconstruction of complex clinical tasks into discrete, teachable components.

This presentation describes the use of simulation technologies as part task trainers within an audiology programme to support the structured development of clinical skills. Simulation equipment is used to target specific competencies, with progressive integration of components as students advance. Skills development areas include case history and feedback, audiological testing and masking, electrophysiological assessments, and real ear measures for hearing aid fitting.

The presentation demonstrates how diverse simulation tools and technologies are strategically employed to support skill acquisition within these domains. The overall goal is to combine these components into an integrated simulated case, using established simulation principles, to enable holistic and authentic audiology training once all foundational elements are in place.



Baby Isao with the appropriate electrode setup for the testing. Students will be asked to suspend realism and assume the baby is asleep (as this is the ideal for electrophysiology testing), even though she cannot close her eyes.

Simulator modality screen on the facilitator’s laptop. The waveform that is to be expected can be compared to the actual testing waveform that the student is getting from the connection to the Corona EP3 equipment .

Perceived Time vs Objective Time in Audiometric Testing: Fast Bayesian vs Hughson-Westlake Methods in Young Musicians

Ines Ben Aïssa (L3S-ENIT)*; Rim Amara (INSAT & L3S-ENIT); Mériem Jaïdane (dB.Sense & L3S-ENIT)

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Poster

Abstract

This study investigates perceived vs. objective duration in audiometric testing by comparing a Bayesian Active Learning (BAL) approach [1] with the conventional Hughson-Westlake (HW) method in young musicians. Musicians are a particularly relevant population due to their finer pitch discrimination (4-6x) [2] which reduces inter-individual variability and highlights fundamental perceptual mechanisms.

Six young musicians completed two assessments (BAL n=3, HW n=3) separated by a music listening session, with post-session questionnaires on fatigue, perceived duration, and interest in self-administered screening. BAL sampled frequencies continuously across 125-8000 Hz, whereas HW tested 9 standard frequencies with the 2-out-of-3 rule. Median test duration was 5:38 min for BAL vs. 15:52 for HW. Reported fatigue remained low in both groups (BAL: 1/5, HW: 2/5).

Key result: BAL participants accurately estimated test duration (median ratio 1.1) while HW participants substantially underestimated it (2.3). This is consistent with predictability-driven temporal distortion. Indeed, BAL audiometry dynamically adapts stimuli via Gaussian Process inference to select the optimal frequency-level pair at each trial, reconstructing a continuous audiogram while increasing patient throughput. BAL gives an impression of a fully randomized structure in both timing and frequency [3]. In contrast, HW is based on a presentation that is inherently more structured than the BAL protocol: it follows the “down 10 dB / up 5 dB” rule, which produces a globally predictable sequence that engages anticipation mechanisms and compresses perceived duration [4].

We formalize this by treating the sequence of tones generated by both protocols as dynamical systems and computing the maximum Lyapunov exponent of their stimulus sequences as a metric of the predictability horizon [5], using both real experimental data and a virtual patient framework [6] to generate and analyze the corresponding trajectories.

References

[1] Schlittenlacher J, Turner RE, Moore BCJ (2018). Audiogram estimation using Bayesian active learning. J. Acoust. Soc. Am. 144, 421–430.

[2] Micheyl C, Delhommeau K, Perrot X, Oxenham AJ (2006). Influence of musical and psychoacoustical training on pitch discrimination. Hearing Res. 219(1-2), 36–47.

[3] Marin N, Gérenton G, Jean H, Paraouty N, Wallaert N, Lazard DS, Doelling KB, Arnal LH (2025). Predictable sequential structure augments auditory sensitivity at threshold. iScience 28(3), 112074.

[4] Pariyadath V, Eagleman D (2007). The effect of predictability on subjective duration. PLoS ONE 2(11), e1264.

[5] Boffetta G, Cencini M, Falcioni M, Vulpiani A (2002). Predictability: a way to characterize complexity. Phys. Rep. 356(6), 367–474.

[6] Gaied S, Mechergui N, Amara R, Jaïdane M (2025). Exploring the standard Hughson-Westlake protocol in tone audiometry using Gaussian process regression: a virtual patient framework. VCCA.

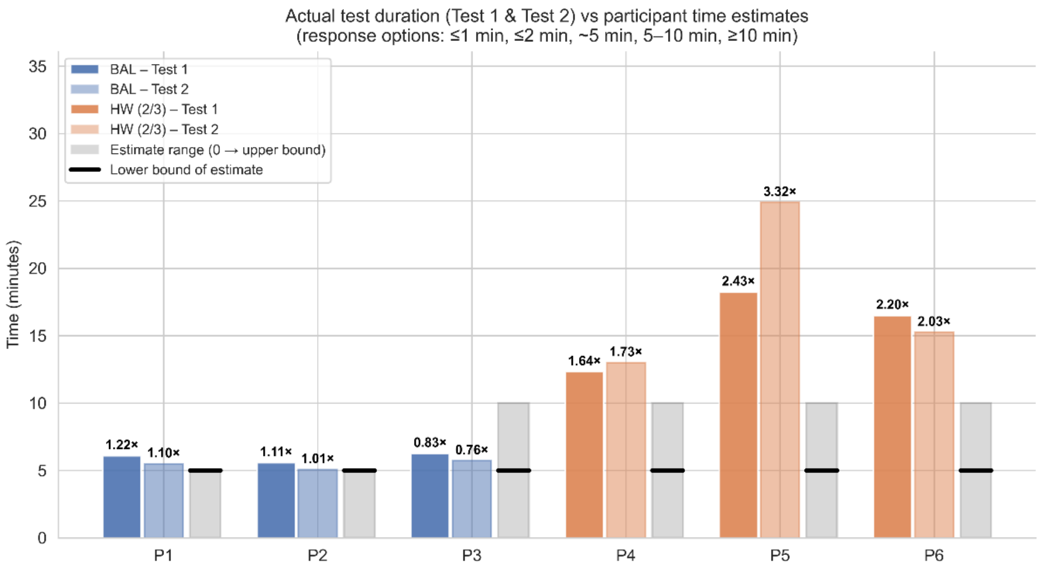

Figure 1 Actual test durations (Test 1 and Test 2) alongside time estimates for six young musicians assigned to either a Bayesian Active Learning (BAL, blue, n=3) or Hughson–Westlake (HW 2/3, orange, n=3) audiometric protocol. Grey bars represent each participant's estimation range from 0 to the upper bound of their selected response category; the black horizontal line marks the lower bound. Ratio values above each bar indicate actual duration divided by estimated midpoint (ratio > 1 = underestimation). Despite HW tests lasting up to 25 minutes, all participants, regardless of protocol, estimated their test duration at either "~5 min" or "5-10 min."

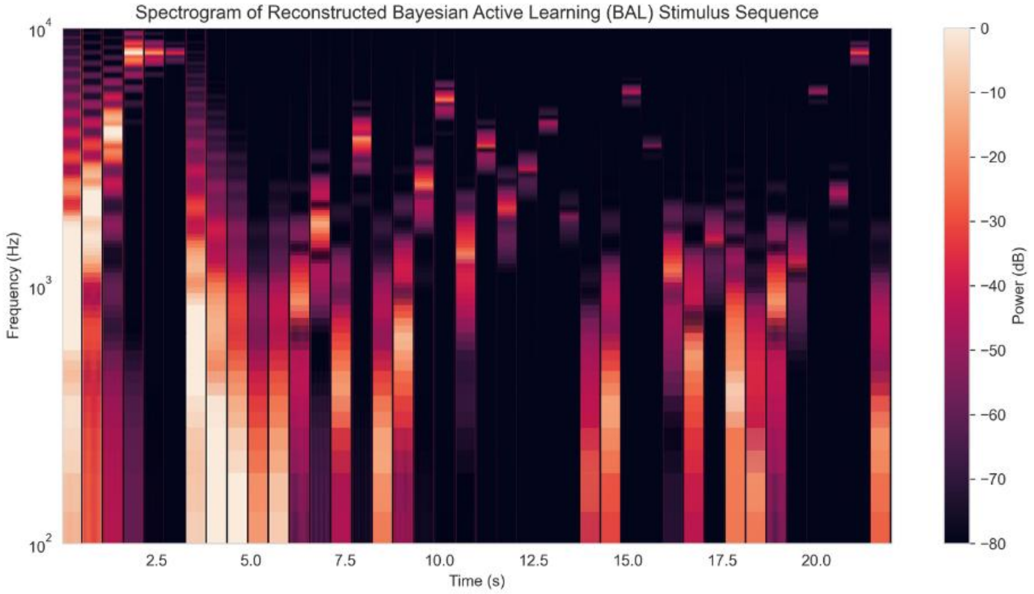

Figure 2a Scattered distribution of energy across frequencies and absence of stable temporal structure (BAL method: adaptive stimulus frequency/level selection based on expected information gain, resulting in a non-sequential exploration of the frequency domain).

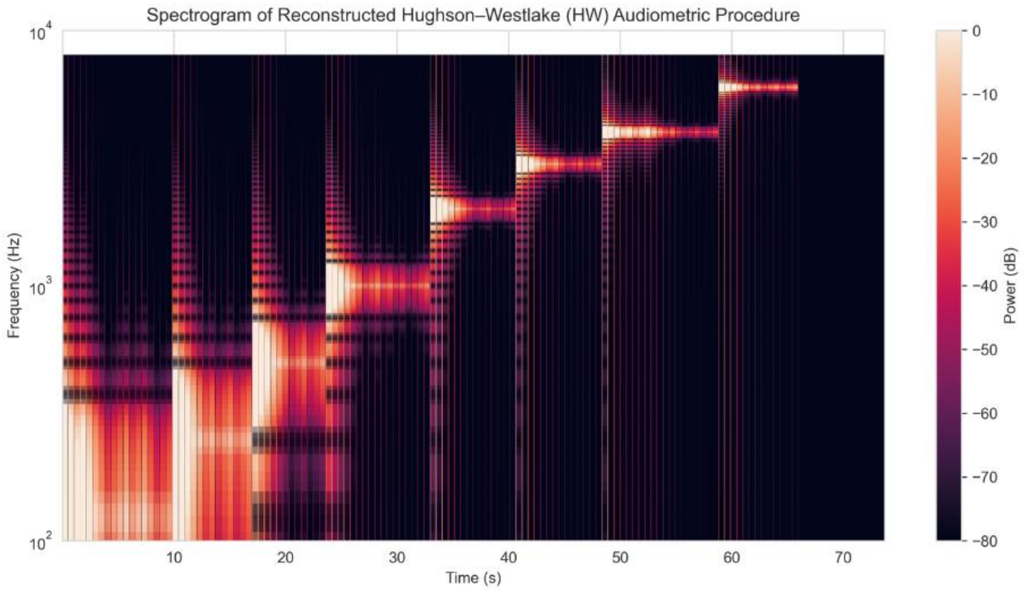

Figure 2b Frequency-specific patterns, reflecting the "deterministic", structured and stepwise nature of the conventional method (HW: Sequentially testing frequencies with a fixed up–down (−10/+5 dB) rule and a 2-out-of-3 response criterion).

Figure 2 Spectrograms of reconstructed stimulus sequences for the Bayesian Active Learning (BAL, Fig. 2a) audiometric procedure and a simulated Hughson–Westlake (HW, Fig. 2b) procedure.
BAL data were obtained from audiometric testing conducted on six young musicians. The HW sequence was generated using a virtual patient model [6] following the standard procedure, providing a controlled reference for comparison.

Session 1.B. Binaural Hearing & Spatial Sound Perception

Localization, spatial cues, and listening in complex acoustic environments.

Chaired by Dr. Jaime Andres Undurraga Lucero.

Contralateral sound attenuation can help hearing-aid users understand speech in realistic, dynamic "cocktail party" listening situations.

Paula García Zaballos (University of Salamanca)*; Enrique A. Lopez-Poveda (University of Salamanca); Almudena Eustaquio-Martín (University of Salamanca); Milagros J. Fumero (University of Salamanca)

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Featured talk

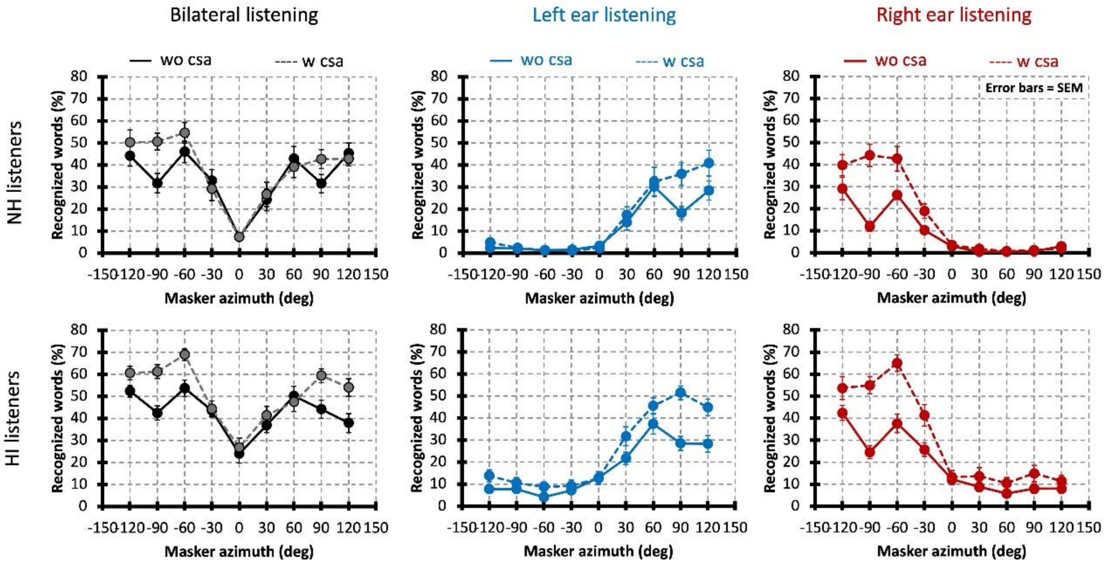
Abstract

Background. Hearing aid (HA) users often struggle to understand speech in “cocktail-party” situations. We have previously shown that contralateral sound attenuation (CSA) (Lopez-Poveda et al., 2022, Hear Res 409:108469) improves speech recognition in free-field conditions. In the present study, we investigated the benefits of CSA in more realistic listening environments.

Method. Eighteen bilateral HA users and ten normal-hearing (NH) listeners were asked to recognize keywords in sentences presented with five simultaneous maskers. Stimuli were delivered via a loudspeaker ring in a low-reverberation room (T60 ~ 0.2 s). The target (female voice, 65 dB SPL) was placed at 0°. Four fluctuating female maskers (60 dB SPL) were positioned symmetrically at ±45° and ±135°. A fifth speech-shaped noise (SSN) masker had its level individually adjusted based on each participant’s speech reception threshold (SRT, 60% correct in S0N300). Its position varied randomly within ±120° across trials to prevent spatial predictability. Participants performed the task listening with the left ear alone, the right ear alone, or both ears. Real-life sounds were captured with an acoustic manikin and sent to listeners in a separate room. They were either left unprocessed or processed with the CSA algorithm and a software hearing aid applying individualized nonlinear, frequency-specific amplification.

Results. CSA significantly improved speech recognition in HA users during bilateral ($p \leq 0.001$), left-ear ($p \leq 0.001$), and right-ear listening ($p \leq 0.001$). NH listeners also showed benefits in bilateral ($p = 0.033$), left-ear ($p = 0.014$), and right-ear conditions ($p \leq 0.001$) (see Figure; García-Zaballos, 2026). The benefit of CSA increased when the SSN masker was located farther from the target speech.

Conclusions. Contralateral sound attenuation improves speech intelligibility in both HA users and NH listeners in realistic, reverberant, and dynamically changing multitalker environments.



The figure shows the percentage of word recognition with and without the CSA processor under bilateral and unilateral listening conditions (left and right ear), comparing normal-hearing participants (upper panels) with participants with hearing loss (lower panels) (García-Zaballos, 2026).

Longitudinal Training Enhances Interaural Timing Sensitivity in Cochlear Implant Users

Raymond Goldsworthy (University of Southern California)*

rgoldswo@usc.edu

Podium

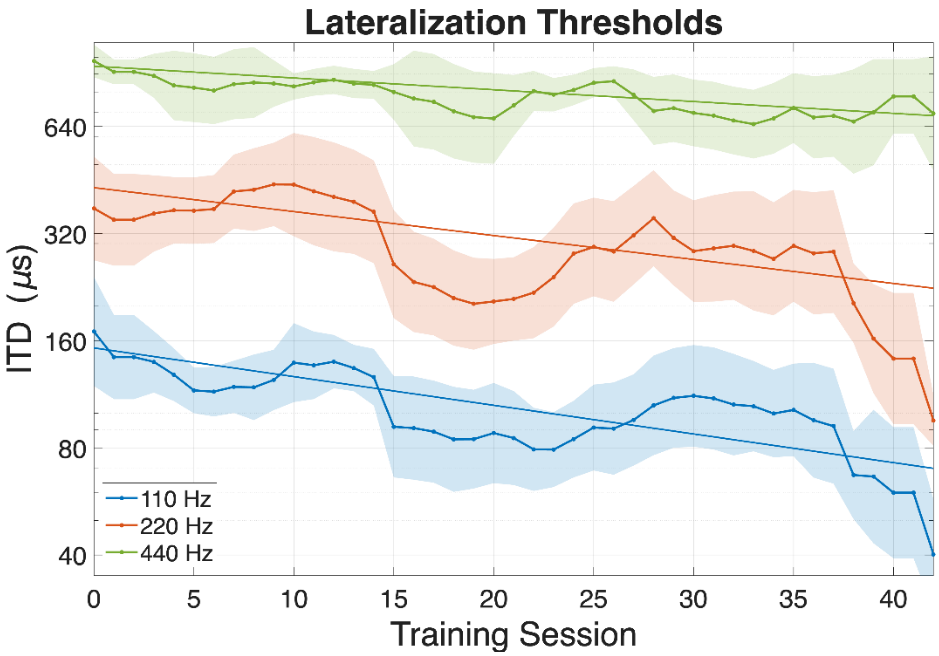
Abstract

Background. Bilateral cochlear implant (CI) users typically show poor sensitivity to interaural time differences (ITDs), in part due to degraded temporal fine-structure (TFS) encoding and limited exposure to consistent binaural timing cues. While laboratory systems can restore precise interaural synchronization, the extent to which users can learn to utilize these cues over time remains unclear.

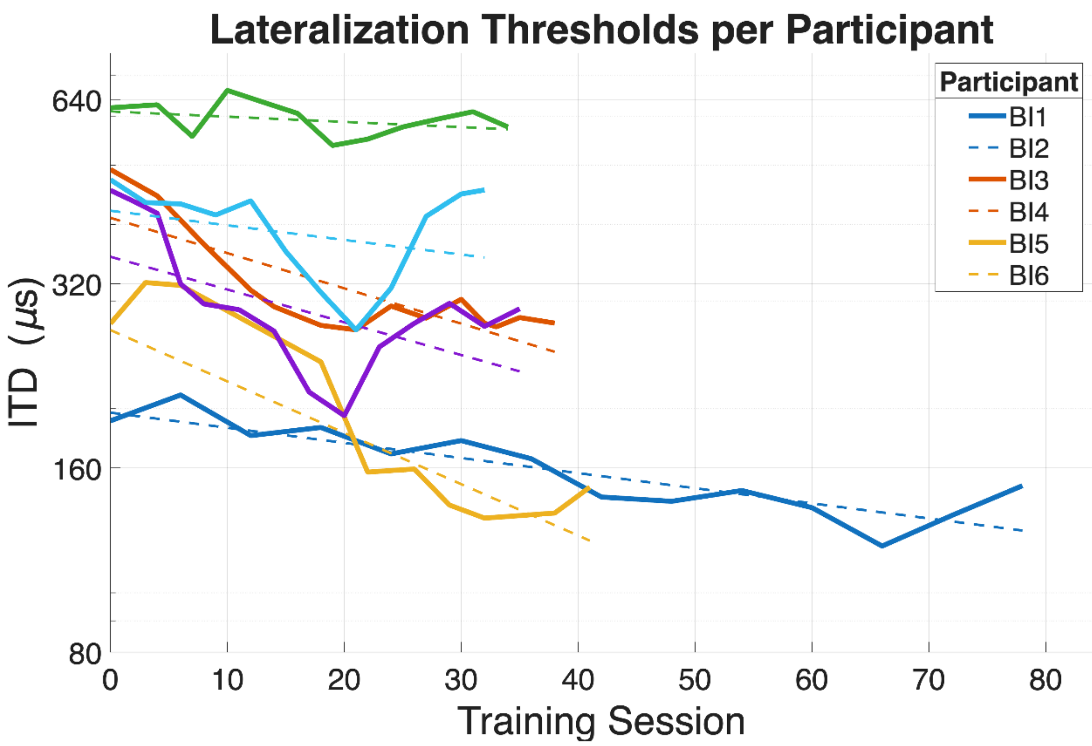
Methods. We evaluated ITD sensitivity in bilateral CI users during a multi-week training protocol using synchronized research processors. Participants completed repeated adaptive ITD discrimination assessments at 110, 220, and 440 Hz without feedback, interleaved with training blocks at a single pulse rate with feedback and performance-based progression. Thresholds were tracked longitudinally across sessions, and changes were analyzed as a function of training exposure and stimulation rate.

Results. ITD discrimination thresholds decreased systematically with training, with the largest and most consistent improvements observed at 110 Hz. Mid-rate stimulation (220 Hz) showed more variable and condition-specific gains, while high-rate stimulation (440 Hz) demonstrated minimal improvement. Learning trajectories extended over many sessions, with gradual reductions in threshold and partial generalization across conditions.

Conclusions. Extended exposure to synchronized TFS cues can enhance binaural temporal sensitivity in CI users, particularly at low stimulation rates. The results highlight the importance of both stimulus design and training structure in enabling access to binaural timing cues, and suggest that perceptual limitations may reflect insufficient experience as much as encoding constraints. These findings support the integration of structured, rate-specific training paradigms into future CI signal processing and rehabilitation strategies.



Lateralization Thresholds
ITD (μs)
640
320
160
80
40
0
5
10
15
20
25
30
35
40
Training Session
110 Hz
220 Hz
440 Hz



Lateralization Thresholds per Participant
ITD (μs)
640
320
160
80
0
10
20
30
40
50
60
70
80
Training Session
Participant
BI1
BI2
BI3
BI4
BI5
BI6

Headphone-based Spatial Perception Tests in Listeners with Sensorineural Hearing Loss: Insights from Receiver Operating Curve (ROC) diagnostics

P Prameela (AIISH)*; Nisha KV (AIISH); Ajith Kumar Uppunda (AIISH)

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Podium

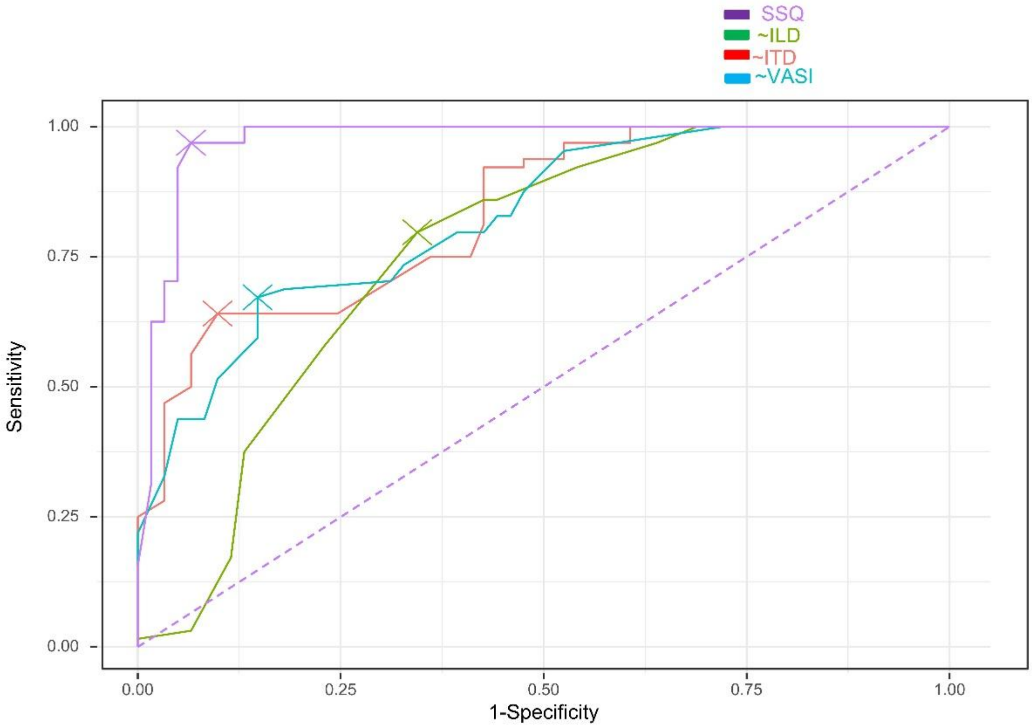
Abstract

Background. Spatial hearing deficits and its ramifications in daily listening conditions is a common complaint encountered by individuals with sensorineural hearing loss (SNHL). There are several spatial hearing test procedures such as free-field loudspeaker based testing, interaural time difference (ITD), interaural level difference (ILD), virtual acoustic space identification (VASI), and the subjective rating questionnaires are available. However, their use in clinical setups is limited, as diagnostic accuracy of these measures are not yet established. The current study was carried out to establish the diagnostic accuracy, and arrive at cut-off criterion for four closed-field spatial hearing tests in SNHL population.

Methods. The study was conducted on 125 individuals (61 SNHL; 64 normal hearing - NH). In phase I, closed-field spatial hearing test battery consisting of ITD, ILD, VASI, and the spatial domain of the Speech, Spatial and Qualities of Hearing Scale (SSQ) were administered. The receiver operating curve (ROC) was utilized to determine sensitivity, specificity, and cutoffs (Youden's index). Phase II involved applying ROC-derived cut-off values to a new sample of 19 participants for group classification.

Results. Based on the ROC analysis, it was found that both SSQ and VASI performed well with regards to their sensitivity, specificity, and the area under the curve. On the other hand, both ITD and ILD tests showed poor performance in differentiating SNHL fron NH group. The reliability of these derived cutoff values was further proven to be 86.7 % effective using the validation data set.

Conclusions. VASI and spatial-SSQ demonstrated high diagnostic accuracy for distinguishing SNHL from NH. Given that VASI is an objective measure, it is recommended as the primary tool for spatial hearing assessment in closed-field. Spatial-SSQ may serve as a complementary test to capture perceptual outcomes.



SSQ
~ILD
~ITD
~VASI
Sensitivity
1-Specificity
1.00
0.75
0.50
0.25
0.00
0.00
0.25
0.50
0.75
1.00

Binaural Neural Processing in Children With and Without a History of Otitis Media

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Podium

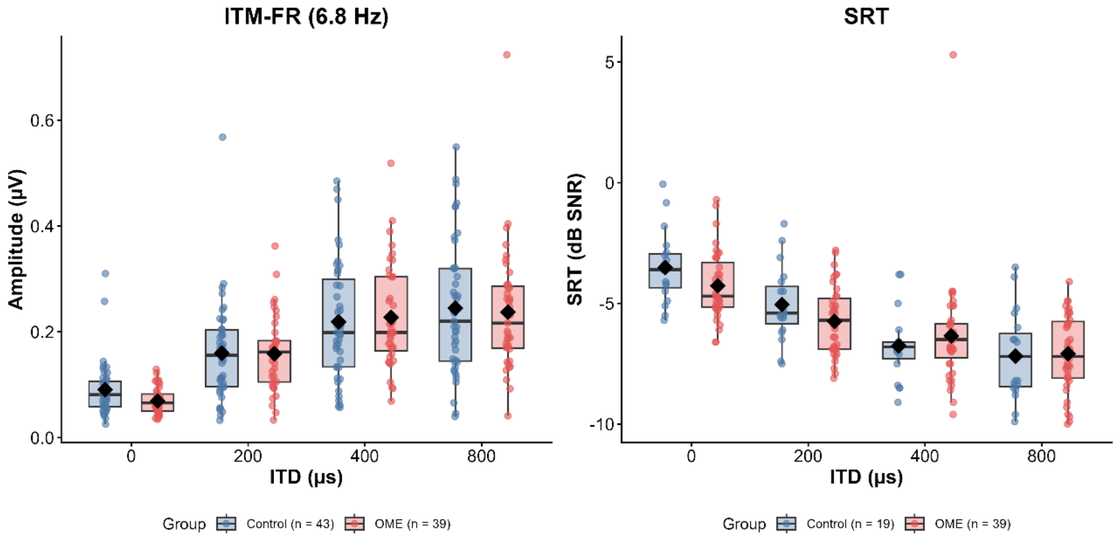
Abstract

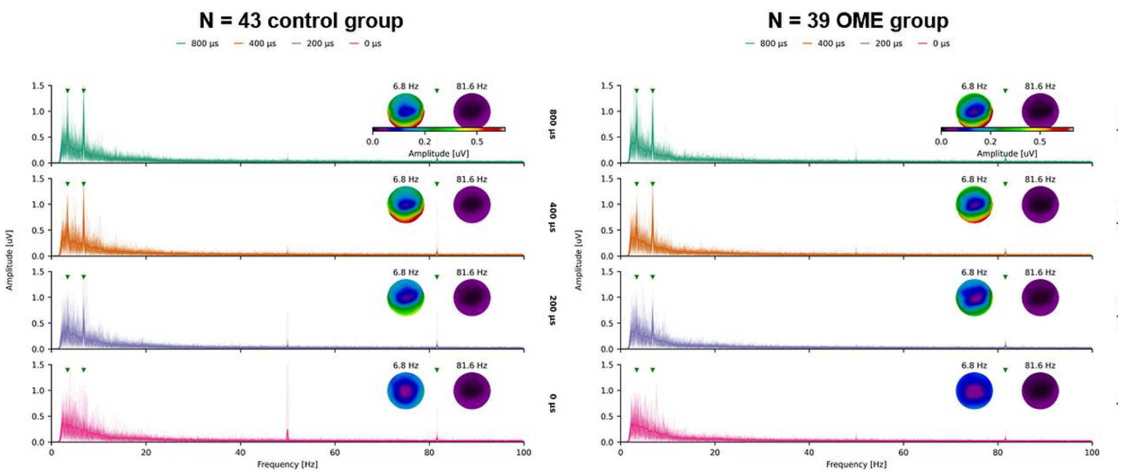
Background. Differences in the arrival time of sounds at the two ears—interaural time differences (ITDs)—enable sound localization and speech perception in complex listening environments, such as noisy classrooms. The brain's ability to process ITD can be disrupted by recurrent episodes of otitis media (OM) during early childhood—a critical period for auditory development. Current assessments of ITD sensitivity in children primarily rely on psychoacoustic measures that are influenced by attention, motivation, and cognition. An objective neural measure of ITD could provide an alternative. Here, we employ the interaural time modulation following response (ITM-FR) to objectively evaluate binaural unmasking in children with or without a history of OM.

Methods. A total of 82 normal-hearing children aged 6–12 years were included: 39 with a medically documented history of OM and 43 controls with no parent-reported history of OM. Neural sensitivity to ITDs was assessed using the ITM-FR, elicited by periodic ITD transitions changing between diotic and dichotic (ITD of 200, 400, or 800 µs) presentation. Using the same ITDs, binaural unmasking was assessed behaviourally with speech reception thresholds (SRTs) measured with a paediatric speech-in-noise test.

Results. Preliminary analyses show consistent effects for both neural and behavioural measures, with larger neural responses and improved SRTs for larger ITDs. For the behavioural measure, a significant ITD-by-group interaction is observable, with no significant post hoc group differences after correction for multiple comparisons.

Conclusion. Larger ITDs are associated with larger neural responses and lower (better) SRTs. No clear group differences are observable using binary OM classification of the participants. Future analyses will examine the effects of the individual OM history (e.g., type, age of onset and offset, number of episodes, and grommet history) on neural and behavioural outcomes.





Speech and Spatial Feature Encoding in Binaural, Naturalistic Audio Representation Models

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Podium

Abstract

Background. Although neural network models for audio are increasingly used as models of auditory processing, they typically lack ecological validity as they address either acoustic processing (e.g., sound classification) or spatial processing (e.g., sound localisation), but not both. To address this, we investigated simultaneous encoding of speech and spatial features in a binaural general-purpose audio representation model capable of performing a wide range of listening tasks, similar to the human auditory pathway.

Methods. We utilised GRAM, a self-supervised model trained on binaural spectrograms of naturalistic sound scenes with reverberation and background noise. We evaluated layer-wise performance on two keyword classification tasks and a sound localisation task. We then used explainability techniques (EigenGradCAM) to identify which spectral regions drive such classification- and localisation decisions.

Results. Layer-wise analysis showed localisation performance peaks at intermediate layer 7, whereas keyword classification peaks at final layer 12, as expected. The explainability analysis revealed a broad cluster (0.4–1.3 kHz) correlating positively with classification performance, while very low (<120 Hz) and high-frequency regions (2.2–16 kHz) correlated negatively. In naturalistic conditions the model narrowed its focus (positive: 0.5–0.7 kHz; negative: <70 Hz and 2.1–3 kHz). For localisation, model focus at 5.8–6.7 kHz correlated positively with performance, while 130–450 Hz and 2.4–3.1 kHz correlated negatively.

Conclusions. Encoding of spatial features occurs earlier in a spatial general-purpose audio representation model than encoding of speech features. Future work will compare GRAM audio representations to measurements of neural activity in the human auditory pathway to assess whether similar trade-offs exist in the brain.

Layer-wise downstream task performance - GRAM-T-Patch

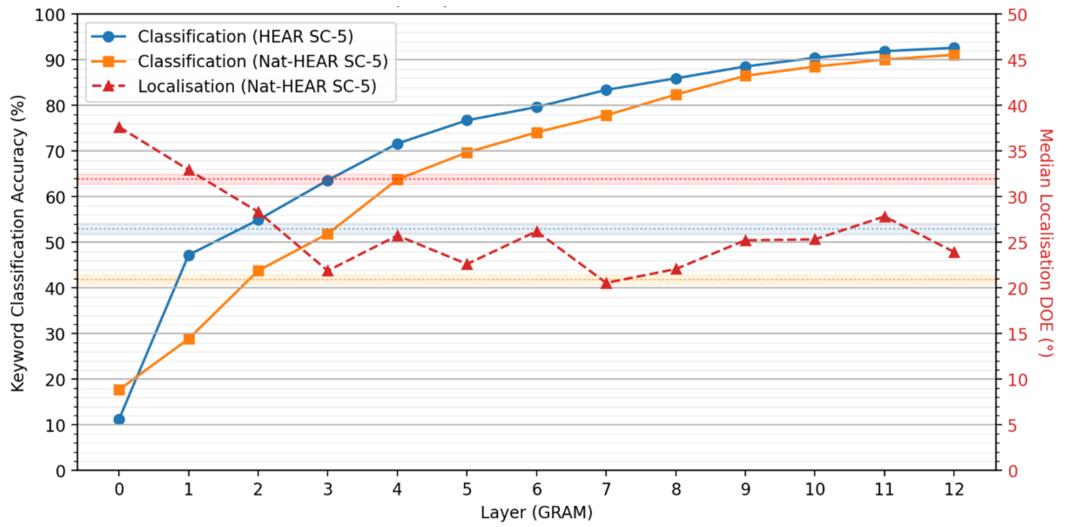
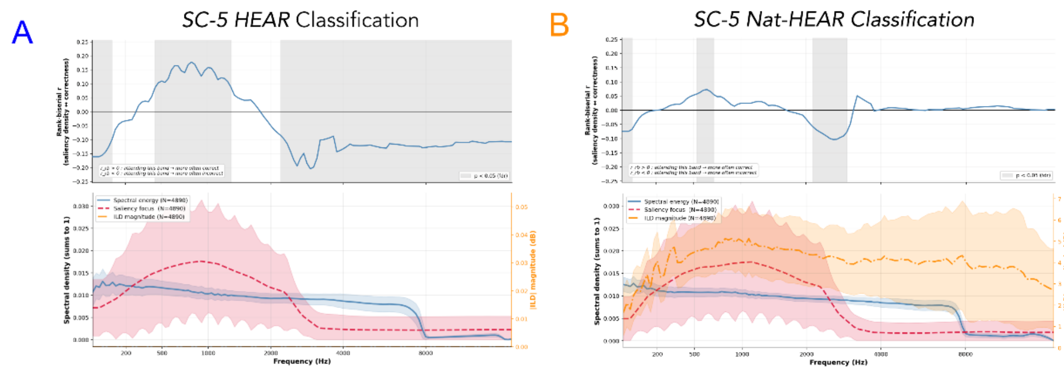


Figure 1. *Layer-wise downstream task performance of our binaural GRAM-T-patch model across 12 model layers. Layer 0 corresponds to raw mel spectrograms from GRAM's feature extractor, prior to any learned transformation. Left y-axis shows classification accuracy (%) for monaural HEAR SC-5 keyword classification (blue) and binaural Nat-HEAR SC-5 keyword classification (orange); right y-axis shows median Direction of Arrival Estimation error (DOE°) for binaural Nat-HEAR SC-5 cartesian regression localisation (red dashed, note that lower error values indicate better localisation.). Horizontal dashed lines indicate baseline-level performance from 10 randomly initialised GRAM networks, coloured respectively for each task (mean ± 95% CI).*

Correlation experiment - time-averaged EigenGradCAM saliency ~ task performance per mel frequency bin



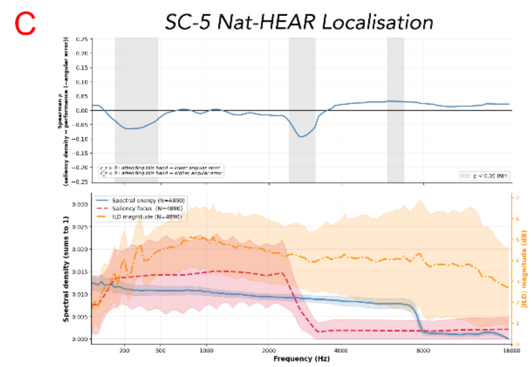


Figure 2. *Time-averaged EigenGradCAM saliency correlated with task performance per frequency bin (N=4980, full test set). Grey shading indicates FDR-corrected significant frequency bins. Lower subpanels show spectral energy (blue) and saliency focus (red dashed) on a shared probability density y-axis, and ILD magnitude (orange) on it's own y-axis, across frequency. All frequency axes are plotted with mel spacing. (A) Wilcoxon rank-sum test between EigenGradCAM saliency and binary classification correctness for the monaural HEAR SC-5 keyword classification task. (B) Same test for the binaural naturalized HEAR SC-5 keyword classification task. (C) Spearman rank correlation between saliency and negated angular error for the binaural naturalized HEAR SC-5 cartesian regression localisation task. In all plots, positve clusters indicate model saliency in these frequency bands improves task performance.*

Perceptual Weighting of Auditory Localization Cues Across Frequency Bands in Free-field

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Podium

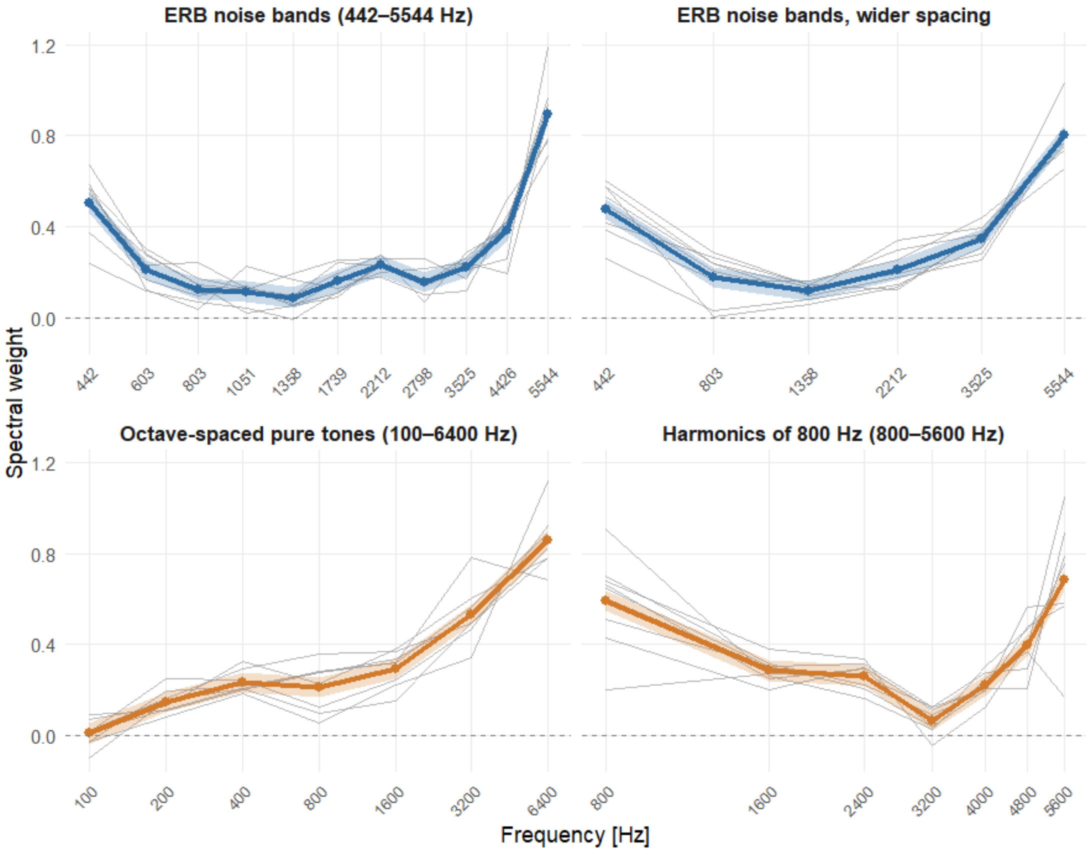
Abstract

Background. The auditory system relies on interaural time and level differences (ITD and ILD) to locate sounds in the horizontal. Spectral weighting functions (SWFs) describe how much each frequency band contributes to the lateralization judgement. Earlier work investigating SWFs for ITDs and ILDs in isolation over headphones showed the lowest and the highest bands receiving highest weights, respectively. Another study employing loudspeaker playback reported a peak near 800 Hz, consistent with an ITD dominance region. It remains unclear how the auditory system integrates information across frequency bands.

Methods. SWFs were measured in normal-hearing listeners using 11 loudspeakers in the horizontal plane located between ±75°. In each trial, several frequency bands were played simultaneously from independently chosen loudspeaker positions, and listeners reported whether they perceived the stimulus on the left or right. Four conditions were tested: 11 ERB noise bands between 442 and 5544 Hz, 6 ERB bands with wider spacing, 7 octave-spaced tones from 100 to 6400 Hz, and 7 harmonics of an 800 Hz tone between 800 and 5600 Hz. A reverse correlation approach was used to estimate the SWF.

Results. In the noise conditions, the highest weights were found for the highest frequency band followed by the lowest band. In the octave-spaced tone condition, only the highest frequency dominated, indicating a one-sided edge effect. The broadband conditions showed no peak around 800 Hz; high weights only appeared when 800 Hz was the lowest band.

Conclusion. The results show that the auditory system appears to weigh spectral edges more strongly when locating a sound in the presence of conflicting spatial information. The band with the strongest contribution to the localization response was found to be at high frequencies, indicating that ILDs contributed more strongly than ITDs. This might explain why people with hearing impairments struggle locating sounds in noisy environments.


ERB noise bands (442–5544 Hz)
ERB noise bands, wider spacing
Octave-spaced pure tones (100–6400 Hz)
Harmonics of 800 Hz (800–5600 Hz)
Spectral weight
Frequency [Hz]
1.2
0.8
0.4
0.0
442
603
803
1051
1358
1739
2212
2798
3525
4426
5544
100
200
400
800
1600
3200
6400
2400
4000
4800
5600

Effect of Binaural Cue Distortions on Music Quality Ratings with Simulated Hearing Loss and Hearing Aids

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Poster

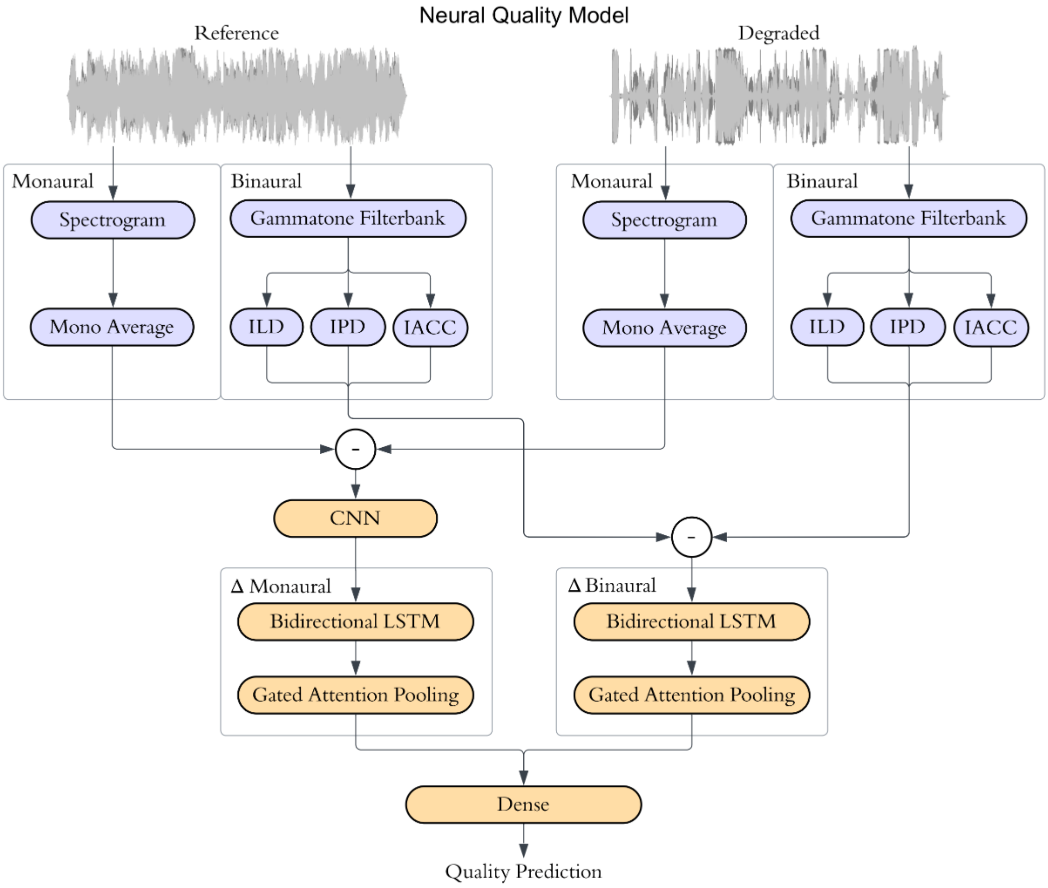
Abstract

Background. Music quality in hearing aids has previously been investigated as a monaural percept, overlooking the potential influence of distortions to interaural cues induced by hearing aid processing. In bilateral fittings, such distortions can disrupt interaural time, level and coherence cues. While these disruptions are known to negatively affect auditory scene analysis, their effect on perceived music quality remains unclear.

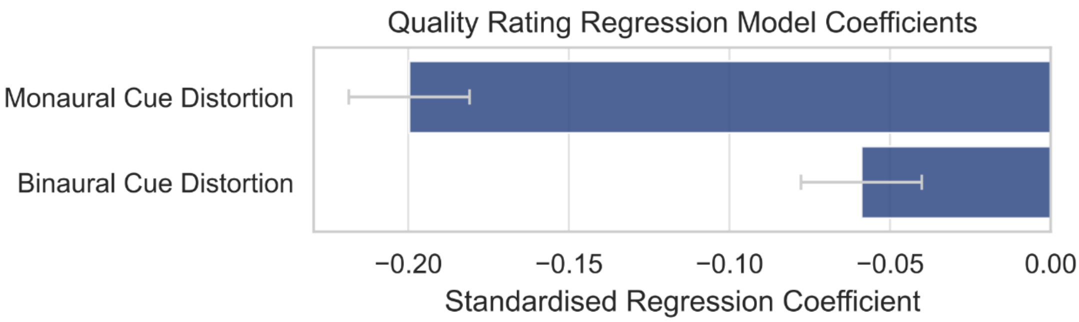
Methods. 268 music signals pre-processed with a hearing aid and hearing loss simulator were assessed by 20 normal hearing participants using an adaptive pairwise comparison paradigm. Estimated quality scores were modelled using features capturing distortions to monaural magnitude spectra and interaural cues. A neural model was also trained to predict perceived quality from combined monaural and interaural features.

Results. Regression analysis showed that binaural cue distortions were associated with a statistically significant decrease in perceived quality, beyond the contribution of monaural spectral distortions. The neural model improved prediction accuracy relative to several existing models but did not exceed channel-wise averaged predictions from the best-performing monaural model. This pattern is consistent with a relatively small effect size of interaural cue distortions. Further analyses suggest that relative weighting of distortion types may vary with the acoustic characteristics of the signal.

Conclusions. These results indicate that binaural cues contribute to music quality with simulated hearing loss and hearing aids; future approaches may benefit from accounting for interaural cue preservation and interactions between signal processing and the acoustic characteristics of the input signal.



Neural Quality Model
Reference
Degraded
Monaural
Binaural
Spectrogram
Gammatone Filterbank
Mono Average
ILD
IPD
IACC
-
CNN
Δ Monaural
Δ Binaural
Bidirectional LSTM
Gated Attention Pooling
Dense
Quality Prediction



Quality Rating Regression Model Coefficients
Monaural Cue Distortion
Binaural Cue Distortion
−0.20
−0.15
−0.10
−0.05
0.00
Standardised Regression Coefficient

Task-Optimized Models of Uncertainty Reproduce Human Confidence Judgments

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Poster

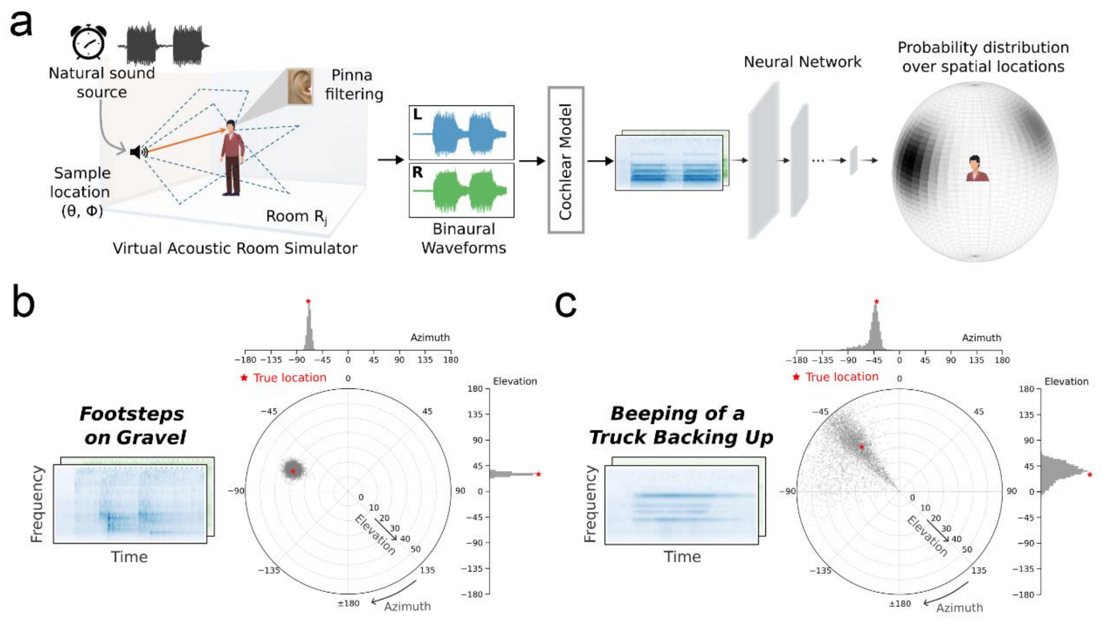
Abstract

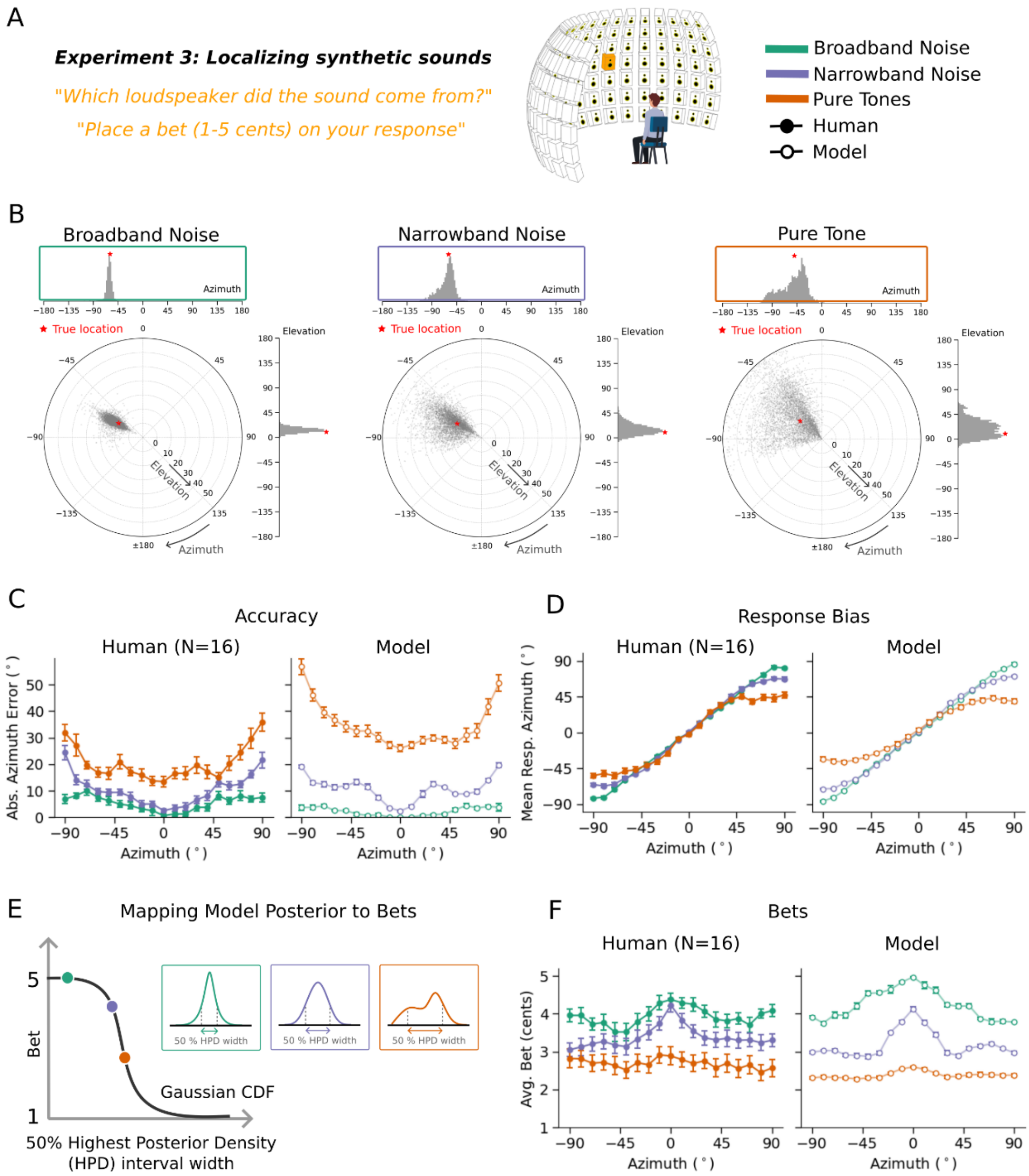
Background. Sensory inferences about the world are made from ambiguous observations and are thus inevitably uncertain. Moreover, uncertainty is likely to be aggravated by hearing loss. Accurate estimation of uncertainty could be critical to decisions about where and when to act, but little is known about how uncertainty is estimated for real-world perceptual problems.

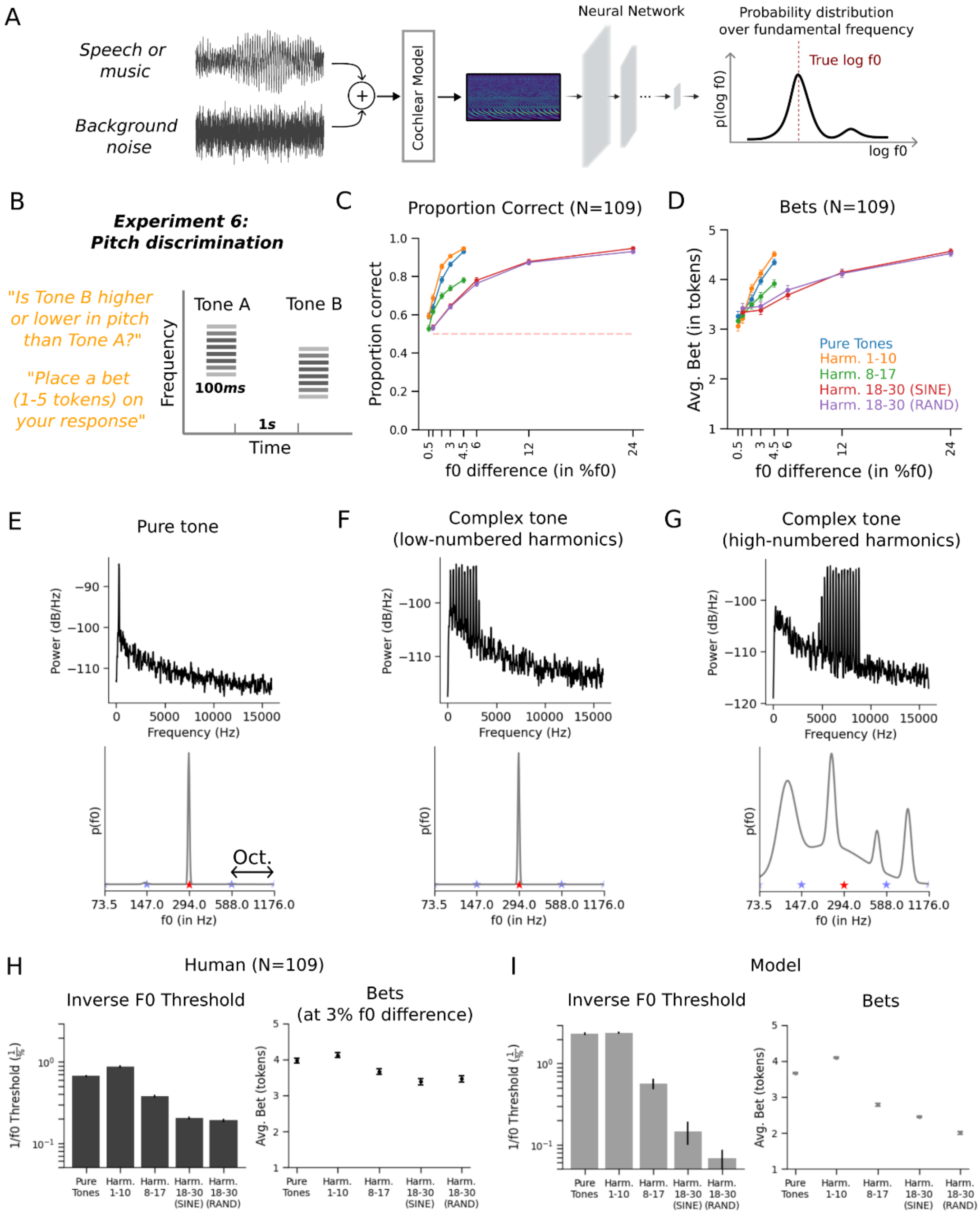
Methods. We developed a new class of stimulus-computable models that explicitly represent uncertainty over perceptual variables and optimized them for sound localization and pitch estimation. The models estimated parameters of a probability distribution over location or fundamental frequency. The models were optimized to maximize the likelihood of ground-truth labels for large datasets of natural sounds, utilizing binaural spatial renderings for localization and short excerpts of speech or music for pitch estimation. The models should learn to produce broader or multimodal distributions for ambiguous stimuli. We compared model uncertainty estimates to human confidence judgments obtained by asking participants to make localization or pitch judgments and place monetary bets (1–5 cents) to indicate confidence. To simulate the same experiments on models, we took a model's confidence to be a summary measure of the spread of the model posterior distribution.

Results. Humans placed lower bets for peripheral locations and narrow-band sounds in localization, and for complex tones with high-numbered harmonics in pitch. These effects were closely mirrored by model predictions across both domains.

Conclusions. Humans have internal estimates of the uncertainty of their auditory percepts of sound location and pitch. These confidence judgments closely match the uncertainty estimates of models optimized for accurate performance in each domain, suggesting that human confidence is normatively appropriate. This framework provides a general approach for investigating confidence across diverse perceptual domains.


a
Natural sound source
Pinna filtering
Sample location (θ, Φ)
Room R_j
Virtual Acoustic Room Simulator
L
R
Binaural Waveforms
Cochlear Model
Neural Network
Probability distribution over spatial locations
b
Footsteps on Gravel
Frequency
Time
Azimuth
True location
Elevation
c
Beeping of a Truck Backing Up
Frequency
Time
Azimuth
True location
Elevation


A
Experiment 3: Localizing synthetic sounds
"Which loudspeaker did the sound come from?"
"Place a bet (1-5 cents) on your response"
Broadband Noise
Narrowband Noise
Pure Tones
Human
Model
B
Broadband Noise
Narrowband Noise
Pure Tone
Azimuth
True location
Elevation
C
Accuracy
Human (N=16)
Model
Abs. Azimuth Error (°)
Azimuth (°)
D
Response Bias
Human (N=16)
Model
Mean Resp. Azimuth (°)
Azimuth (°)
E
Mapping Model Posterior to Bets
Bet
Gaussian CDF
50 % HPD width
50% Highest Posterior Density
(HPD) interval width
F
Bets
Human (N=16)
Model
Avg. Bet (cents)
Azimuth (°)



A
Speech or music
Background noise
Cochlear Model
Neural Network
Probability distribution over fundamental frequency
True log f0
p(log f0)
log f0
B
Experiment 6: Pitch discrimination
"Is Tone B higher or lower in pitch than Tone A?"
"Place a bet (1-5 tokens) on your response"
Tone A
Tone B
Frequency
100ms
1s
Time
C
Proportion Correct (N=109)
Proportion correct
f0 difference (in %f0)
D
Bets (N=109)
Avg. Bet (in tokens)
Pure Tones
Harm. 1-10
Harm. 8-17
Harm. 18-30 (SINE)
Harm. 18-30 (RAND)
f0 difference (in %f0)
E
Pure tone
F
Complex tone (low-numbered harmonics)
G
Complex tone (high-numbered harmonics)
Power (dB/Hz)
Frequency (Hz)
p(f0)
Oct.
f0 (in Hz)
H
Human (N=109)
Inverse F0 Threshold
Bets (at 3% f0 difference)
I
Model
Inverse F0 Threshold
Bets
Avg. Bet (tokens)
Pure Tones
Harm. 1-10
Harm. 8-17
Harm. 18-30 (SINE)
Harm. 18-30 (RAND)

Impact of Misophonic Triggers on Binaural Auditory Integration: Evidence from Dichotic Listening Testing

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Poster

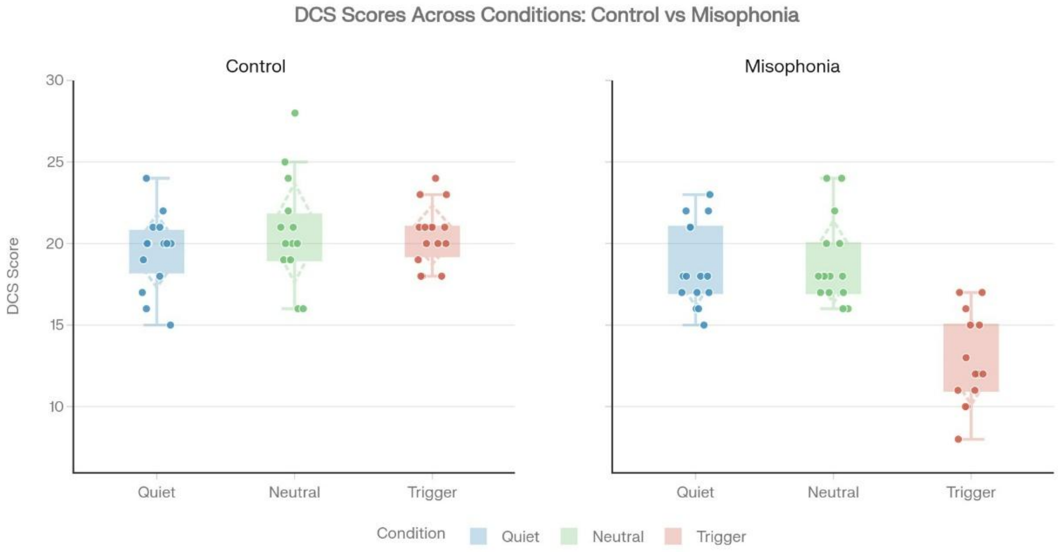
Abstract

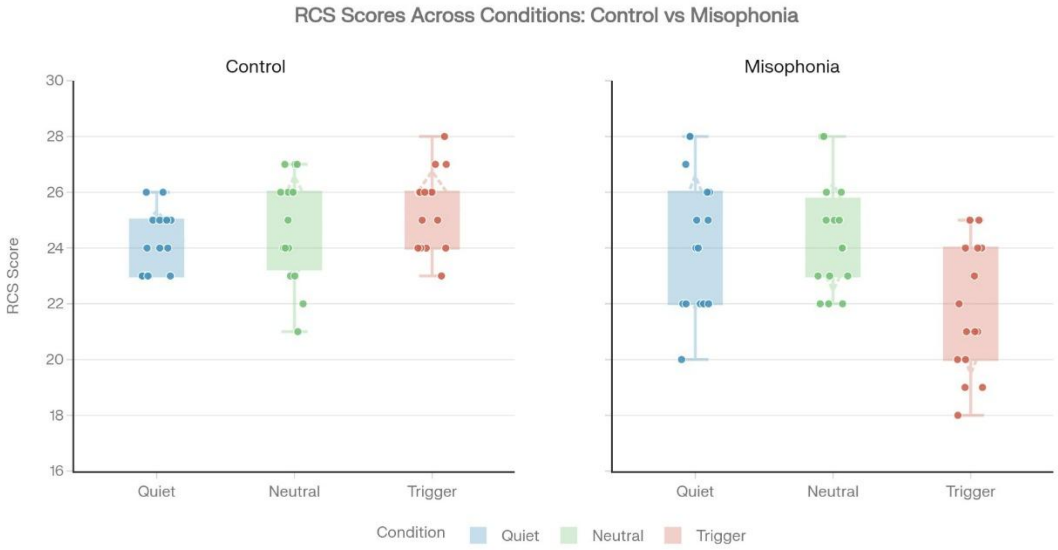
Background. Misophonia is a sound intolerance condition characterized by intense emotional responses to specific repetitive sounds. While increasingly discussed in clinical literature, its status as a distinct audiological disorder remains unresolved. Emerging evidence suggests atypical auditory processing and reduced selective attention during trigger exposure, highlighting the need for objective assessment of central auditory function.

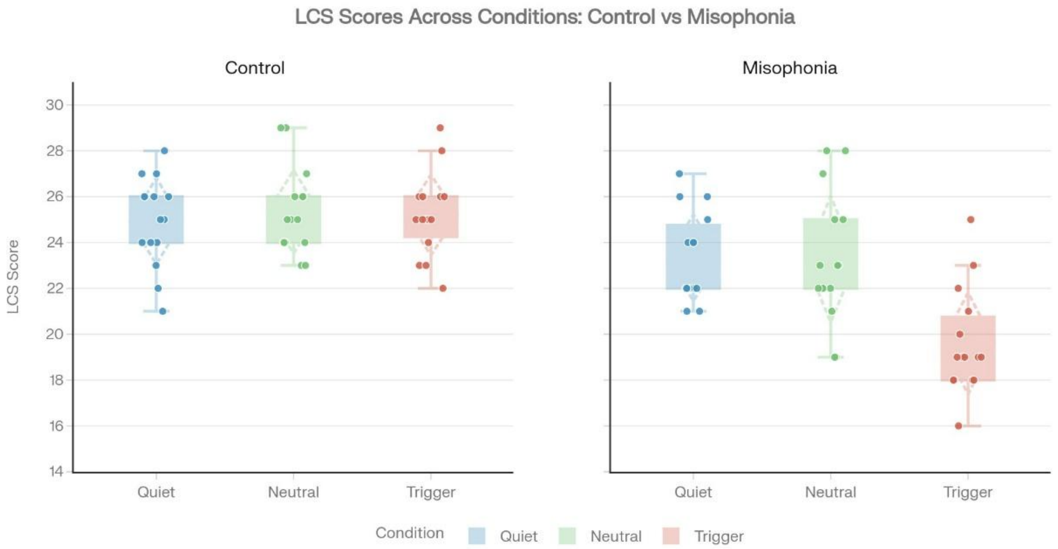
Methods. Fifteen adults with misophonia were compared with age-matched healthy controls. Participants underwent dichotic CV testing under three 1-minute conditions: quiet, neutral, and individualized trigger sounds. Stimuli were drawn from the International Affective Digitized Sounds and validated misophonia sound lists. Right correct scores (RCS), left correct scores (LCS), and double correct scores (DCS) were recorded to assess binaural integration performance.

Results. Control group scores remained stable across all three conditions. In the misophonia group, significant condition effects were observed for RCS ($F = 6.41$, $p = 0.004$, $\eta^2 = 0.23$), LCS ($F = 11.99$, $p < 0.001$, $\eta^2 = 0.36$), and DCS ($F = 26.26$, $p < 0.001$, $\eta^2 = 0.56$), indicating large effect sizes with DCS showing the strongest condition-related deterioration. Between-group comparisons at the trigger condition revealed significantly poorer scores in the misophonia group for RCS ($t = 5.25$, $p < 0.001$, $d = 1.92$), LCS ($t = 7.41$, $p < 0.001$, $d = 2.71$), and DCS ($t = 9.21$, $p < 0.001$, $d = 3.36$), all reflecting very large effect sizes.

Conclusions. Trigger sounds selectively impair auditory listening and binaural integration in individuals with misophonia, while controls remain unaffected. The consistent performance decline across RCS, LCS, and DCS highlights the clinical relevance of incorporating dichotic listening assessment in the audiological evaluation of misophonia.


DCS Scores Across Conditions: Control vs Misophonia
Control
Misophonia
DCS Score
30
25
20
15
10
Quiet
Neutral
Trigger
Condition
Quiet
Neutral
Trigger


RCS Scores Across Conditions: Control vs Misophonia
Control
Misophonia
RCS Score
30
28
26
24
22
20
18
16
Quiet
Neutral
Trigger
Condition
Quiet
Neutral
Trigger


LCS Scores Across Conditions: Control vs Misophonia
Control
Misophonia
LCS Score
30
28
26
24
22
20
18
16
14
Quiet
Neutral
Trigger
Condition
Quiet
Neutral
Trigger

How Does Action-Type Computer Game Playing Affect Hearing?

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Poster

Abstract

Background. Video games and e-sports expose players to high-intensity sounds, which may pose a potential risk for hearing loss. This study aimed to evaluate short-term changes in hearing thresholds and auditory perception before and after gameplay in young adults who regularly play First-Person Shooter (FPS) games including gunfire and explosion sounds.

Methods. This prospective observational study included 33 volunteers aged 18–30 years. The study group consisted of 20 regular gamers, while the control group included 13 non-gamers. Pure-tone hearing thresholds and hearing-in-noise performance were measured at two time points (pre- and post-gameplay) using a mobile hearing test application (e-audiologia.pl) and compared with the control group. Pure-tone screening (0.25–8 kHz) and digit triplet test in noise were conducted using headphones. Additionally, subjective hearing experiences were evaluated using a Visual Analog Scale (VAS).

Results. In the study group, post-game pure-tone averages increased by 2–12 dB. No significant differences were found in pure-tone thresholds between groups ($p > 0.05$), but binaural measurements showed a significant difference ($p = 0.002$). Monaural digit triplet performance ranged from −1.7 to +1.5, indicating variable individual effects. No significant changes were observed in VAS scores ($p = 0.092$).

Conclusion. FPS games may lead to temporary changes in hearing in young adults, even without subjective awareness. Effects were more evident in binaural speech-in-noise performance and high-frequency thresholds. Mobile hearing tests may be useful for monitoring such changes. These findings highlight the importance of considering recreational digital sound exposure as a potential contributor to subclinical auditory changes in daily life. Further studies are required to assess long-term effects.

Keywords: mobile hearing test, noise exposure, FPS games, temporary threshold shift

Influence of Spatial Hearing on Auditory Parallel Subitizing Efficiency

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Poster

Abstract

Background. Subitizing, the rapid and accurate enumeration of small quantities, remains poorly understood in the auditory domain, particularly for simultaneously presented stimuli. Prior work suggested that spatial separation of sound sources may facilitate auditory object individuation and thereby support subitizing. However, it remained unclear whether the benefit arises from alignment with auditory spatial channels or simply from distributing sounds across distinct locations. This study directly compared two spatial configurations to address this question.

Methods. Twenty-four adult participants with normal hearing completed an online enumeration task in which 1 to 6 naturalistic sounds were presented simultaneously via headphones with HRTF-based spatial rendering. In Experiment 1, sounds were distributed across three locations (-90°, 0°, 90°), allowing multiple sounds per location when numerosity exceeded three. In Experiment 2, six distinct locations were used (0°, 60°, -60°, 120°, -120°, 180°), with maxiumum one sound per location. Individual Subitizing Ranges were calculated; accuracy and response times were compared between conditions using paired t-tests.

Results. Participants responded significantly faster in Experiment 1 ($t=2.74$, $p=.012$, $d=0.58$). Accuracy showed a trend toward higher performance in Experiment 2 ($t=-1.97$, $p=.063$, $d=-0.42$), though this did not reach statistical significance.

Conclusions. The results present a mixed picture: greater spatial separation improved accuracy but at the cost of response speed, making interpretation challenging. These findings suggest that spatial distribution of sound sources influences auditory enumeration, though the nature of this relationship requires further investigation. Future studies should employ individualized HRTFs or physical loudspeaker setups to achieve more realistic spatial scenes and allow firmer conclusions about the role of spatial hearing in auditory subitizing.

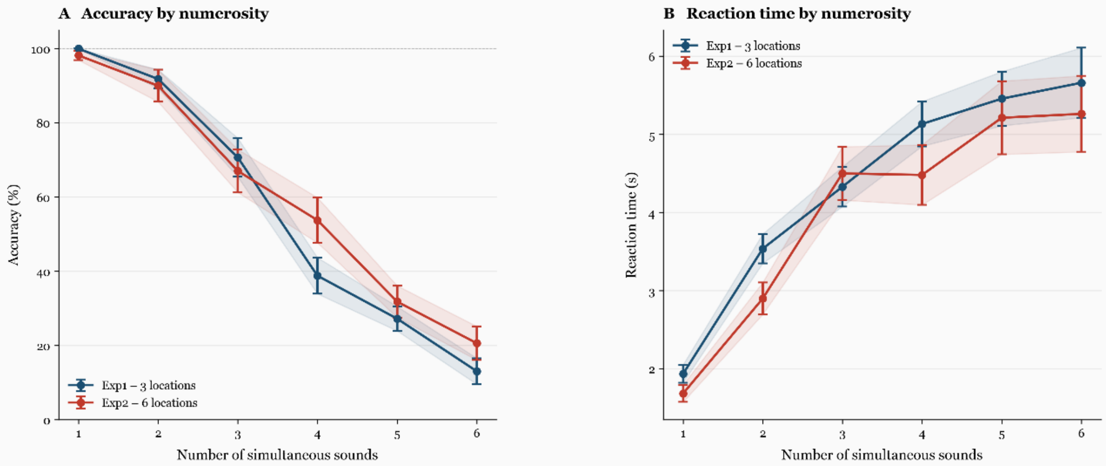

Figure 1. Mean accuracy (A) and reaction time (B) as a function of the number of simultaneous sounds, separately for Exp1 (3 spatial locations) and Exp2 (6 spatial locations). Error bars represent ±1 SE across participants.

Session 2.A. Speech-in-Noise and Auditory Scene Analysis

Perceptual and neural mechanisms supporting speech understanding in real-world conditions.

Chaired by Dr. Miriam I. Marrufo-Pérez.

Attention to Speech Modulates Distortion Product Otoacoustic Emissions Evoked by Speech-Derived Stimuli in Humans

Janna Steinebach (Friedrich-Alexander-University Erlangen-Nuremberg)*; Tobias Reichenbach (Friedrich-Alexander-University Erlangen-Nuremberg)

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Featured talk

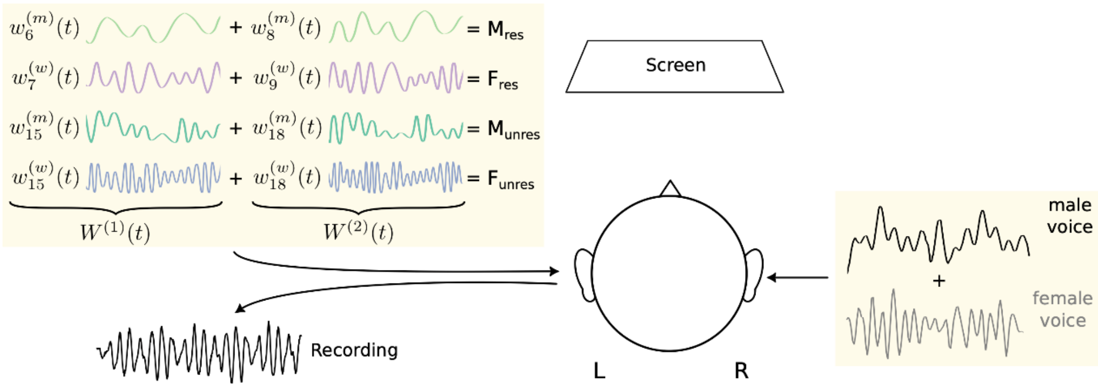
Abstract

Background. Understanding speech in noisy environments requires selective attention to a target talker. Although this ability is largely attributed to cortical processing, descending pathways link higher auditory centers to the cochlea, where an active, frequency-selective amplification process operates. Distortion-product otoacoustic emissions (DPOAEs), generated by this nonlinear mechanism, provide a noninvasive measure of cochlear function. We investigated whether selective and intermodal attention influence cochlear responses using speech-like DPOAEs.

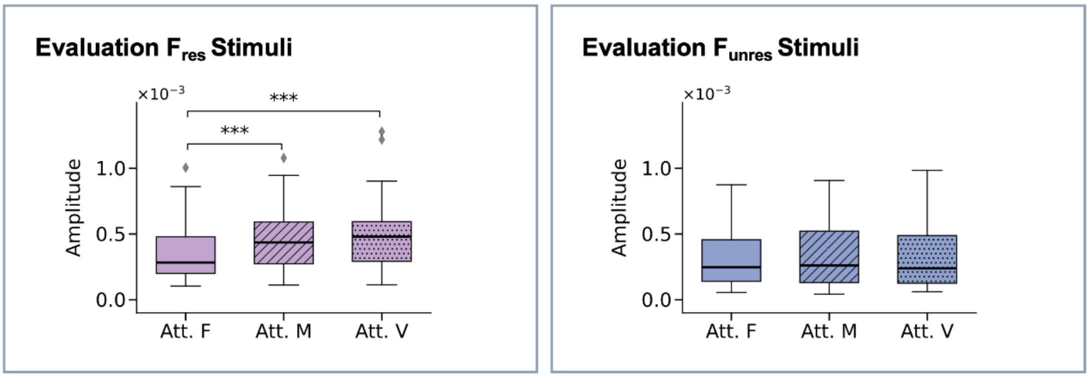
Methods. Speech-like DPOAE stimuli were derived from the harmonic structure of voiced speech and enabled simultaneous assessment of responses to two competing voices. Forty participants listened to concurrent male and female speech streams and alternated attention between the two talkers or a visual distractor. Cochlear responses were recorded from the ear canal in the form of speech-like DPOAEs.

Results. Speech-like DPOAEs linked to spectrally resolved harmonics were significantly reduced when the corresponding voice was attended compared to when it was ignored. No attentional modulation was observed for unresolved harmonics of the target voice when the competing voice contained unresolved components in the same frequency range. Similar effects were found when comparing auditory and visual attention, indicating that intermodal attention also affects cochlear responses.

Conclusions. These findings indicate that selective attention to speech in noise could potentially modulate the cochlear active process. The observed specificity to resolved harmonics suggests that attentional influences at the auditory periphery would largely depend on spectral separability. The speech-like DPOAE approach provides a novel tool for examining peripheral contributions to auditory scene analysis under realistic listening conditions.



$w_6^{(m)}(t)$ + $w_8^{(m)}(t)$ = M_{res}
$w_7^{(w)}(t)$ + $w_9^{(w)}(t)$ = F_{res}
$w_{15}^{(m)}(t)$ + $w_{18}^{(m)}(t)$ = M_{unres}
$w_{15}^{(w)}(t)$ + $w_{18}^{(w)}(t)$ = F_{unres}
$W^{(1)}(t)$
$W^{(2)}(t)$
Screen
Recording
L
R
male voice
+
female voice



Evaluation F_{res} Stimuli
$\times 10^{-3}$
***
***
Amplitude
1.0
0.5
0.0
Att. F
Att. M
Att. V
Evaluation F_{unres} Stimuli
$\times 10^{-3}$
Amplitude
1.0
0.5
0.0
Att. F
Att. M
Att. V

Speech-in-Noise Difficulties in Aminoglycoside Ototoxicity Reflects Combined Afferent and Efferent Dysfunction

Lina Motlagh Zadeh (University of Cincinnati)*; Diala Izhiman (University of Texas at Dallas); Chelsea Blankenship (Cincinnati Children's Hospital Medical Center); David Moore (Cincinnati Children's Hospital Medical Center); Dawn Konrad Martin (National Center for Rehabilitative Auditory Research, VA Portland Health Care System); Angela Garinis (Department of Otolaryngology/Head & Neck Surgery, Oregon Health & Science University); Patrick Feeney (Department of Otolaryngology/Head & Neck Surgery, Oregon Health & Science University); Lisa Hunter (Cincinnati Children's Hospital Medical Center)

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Podium

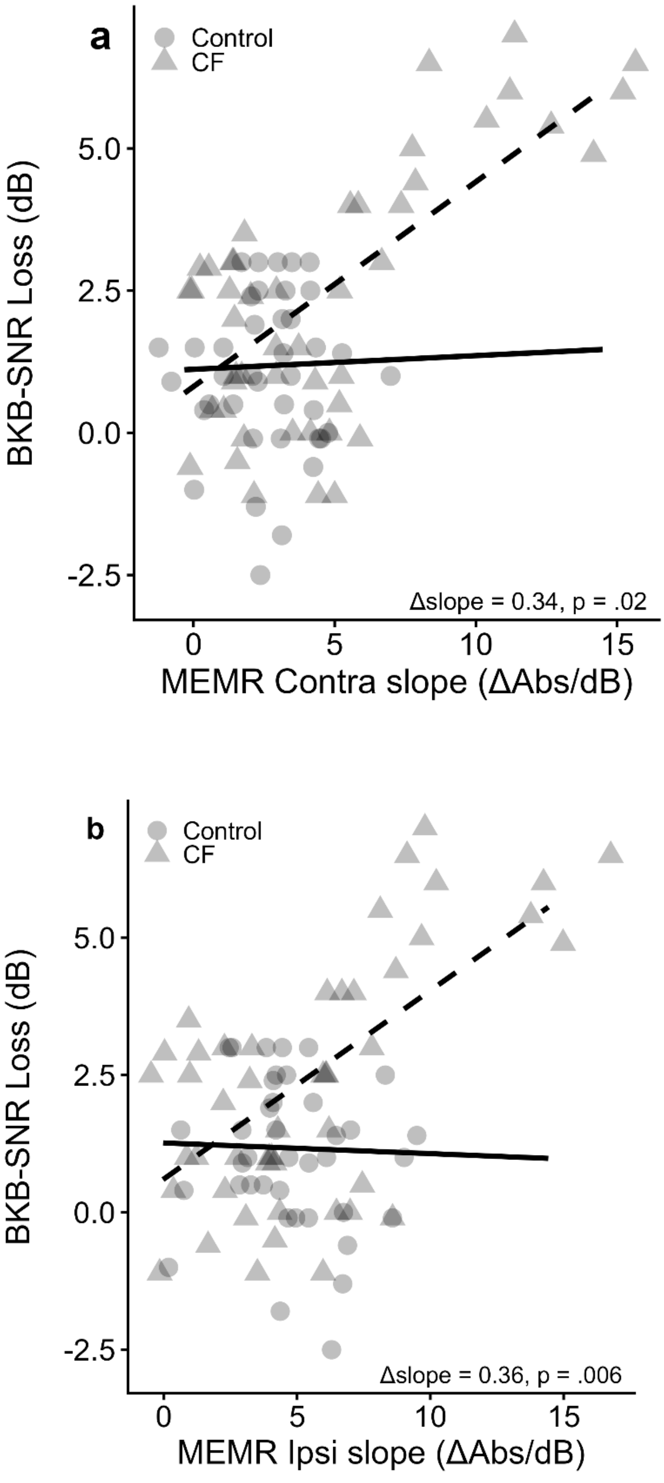
Abstract

Background. Patients with cystic fibrosis (CF) frequently receive aminoglycosides (AGs) to manage recurrent pulmonary infections, placing them at risk for ototoxicity. Chronic AG exposure can lead to cochlear damage affecting hair cells and neurons, with greatest vulnerability in the basal cochlea and potential spread to more apical regions. Although extended-high-frequency (EHF; 9-6kHz) hearing loss is often the earliest indicator of AG ototoxicity, its functional consequences for speech-in-noise (SiN) perception remain unclear. We hypothesized that SiN difficulties in CF individuals treated with AGs reflect combined cochlear and neural dysfunction, primarily involving EHFs with additional contributions from standard frequencies (SF; 0.25-8kHz). Three mechanisms contributing to SiN perception were examined: (1) primary effects of reduced EHF sensitivity, measured by pure-tone audiometry (PTA) and transient-evoked otoacoustic emissions (TEOAEs); (2) secondary effects of subclinical SF damage, measured by PTA and TEOAEs; and (3) neural contributions, assessed via middle-ear muscle reflex (MEMR) thresholds and growth functions.

Methods. A total of 185 participants were enrolled, including 101 individuals with CF treated with intravenous AGs and 84 age- and sex-matched controls without CF or hearing concerns. Assessments included EHF and SF PTA, Bamford-Kowal-Bench (BKB)-SIN testing, TEOAEs (0.71–14.7 kHz), and ipsilateral and contralateral wideband MEMR measures.

Results. Reduced EHF sensitivity (PTA, TEOAEs) was not associated with impaired SiN in the CF group. In contrast, SF hearing was the primary predictor of SiN performance, independent of EHF. Increased MEMR growth was significantly associated with poorer SiN.

Conclusions. SiN deficits in CF are primarily driven by SF impairment, with additional contributions from altered neural reflex function. These findings support integrated sensory and neural mechanisms underlying listening difficulties following AG exposure.



Behavioural and Neurophysiological Correlates of Adaptation to Noise in AM Detection

Iñigo Santamaría Fernández (Universidad de Granada)*; Miriam Marrufo-Pérez (University of Salamanca); Enrique A. Lopez-Poveda (University of Salamanca); Joaquin T. Valderrama (University of Granada)

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Podium

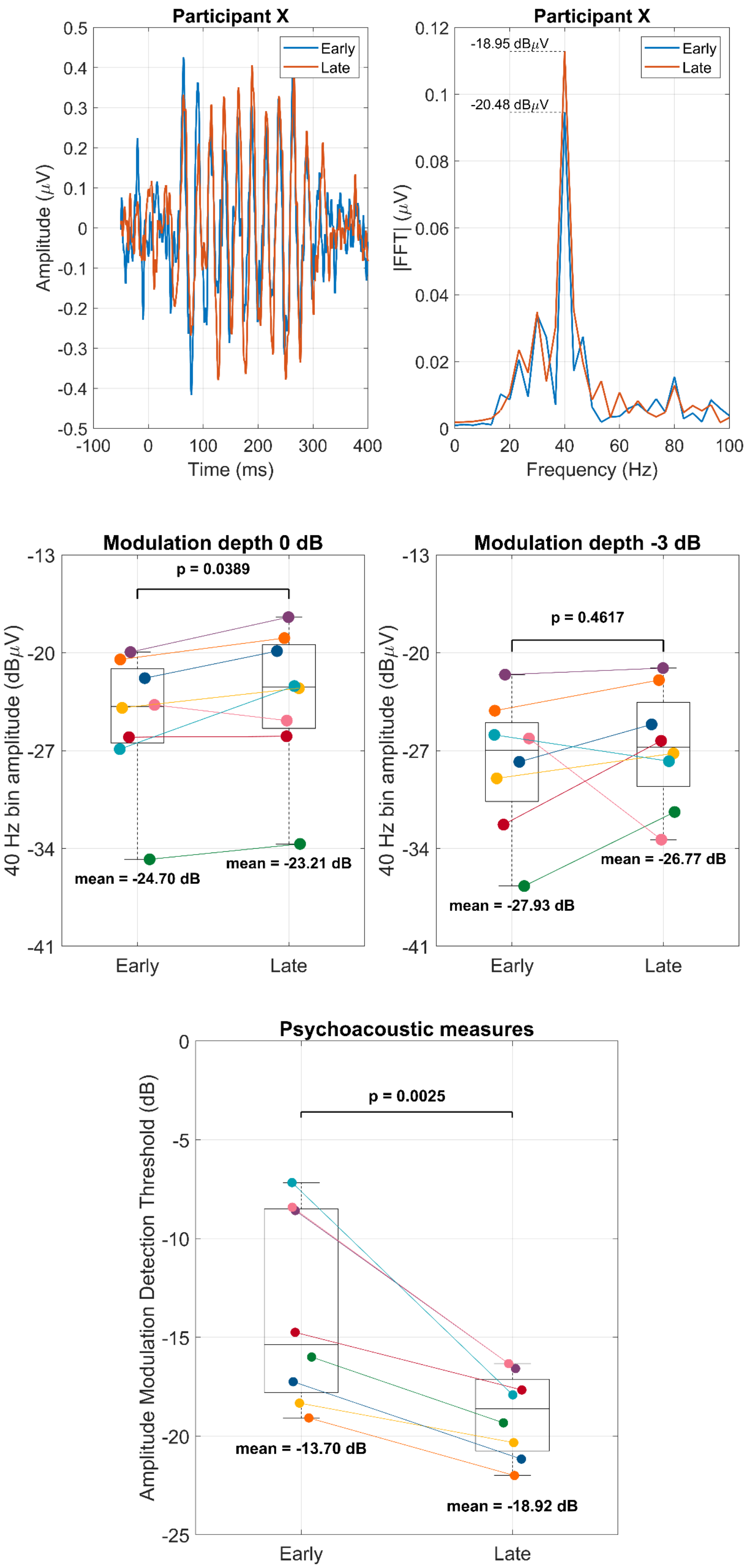
Abstract

Background. "Adaptation to noise" refers to the improvement in detection and recognition of a signal when delayed relative to noise onset. This phenomenon may reflect the auditory system's ability to adjust neural responses to optimise signal encoding in acoustically challenging environments. While adaptation to noise has been extensively investigated behaviourally, physiological evidence of this adaptation remains scarce. This study aimed to investigate neurophysiological markers of adaptation to noise and their relationship with behavioural adaptation.

Methods. Eight normal-hearing adults (4 males; aged 21–40 years; ongoing) participated. AM detection thresholds were measured for tones (modulation frequency = 40 Hz) presented in broadband noise at 0 dB SNR, at two temporal positions relative to noise onset: 50 ms (early) and 500 ms (late), and adaptation was regarded as the AM detection threshold difference between the two conditions. Auditoy steady-state responses (ASSRs) were recorded using identical stimuli and temporal positions, with modulation depths of 0 dB and −3 dB.

Results. AM detection thresholds improved significantly from early to late conditions (mean: −13.70 dB vs. −18.92 dB; $p = 0.0025$; Fig. 3), indicating enhanced AM sensitivity following adaptation. ASSR amplitude at 40 Hz increased significantly from early to late at 0 dB modulation depth (mean: −24.70 dB vs. −23.21 dB; $p = 0.0389$), but not at −3 dB ($p = 0.4617$; Fig. 2). Representative waveforms and FFT spectra illustrating the 40 Hz ASSR across conditions are shown in Fig. 1.

Conclusion. The improvement in AM detection thresholds confirms robust adaptation to noise. ASSRs amplitudes at 40 Hz partially reflected these behavioural findings, though considerable inter-individual variability was observed. These preliminary results suggest ASSRs may serve as an objective measure of adaptation to noise; however, larger samples are needed before firm conclusions can be drawn.


Participant X
Early
Late
Amplitude (μV)
Time (ms)
Participant X
-18.95 dBμV
-20.48 dBμV
Early
Late
|FFT| (μV)
Frequency (Hz)
Modulation depth 0 dB
p = 0.0389
mean = -24.70 dB
mean = -23.21 dB
40 Hz bin amplitude (dBμV)
Early
Late
Modulation depth -3 dB
p = 0.4617
mean = -27.93 dB
mean = -26.77 dB
40 Hz bin amplitude (dBμV)
Early
Late
Psychoacoustic measures
p = 0.0025
mean = -13.70 dB
mean = -18.92 dB
Amplitude Modulation Detection Threshold (dB)
Early
Late

The Effect of Changing Talker Voices on Speech Perception in Individuals with Normal Hearing

Saranya Mundayoor (The University of Texas at Dallas)*; Edward Lobarinas (The University of Texas at Dallas)

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Podium

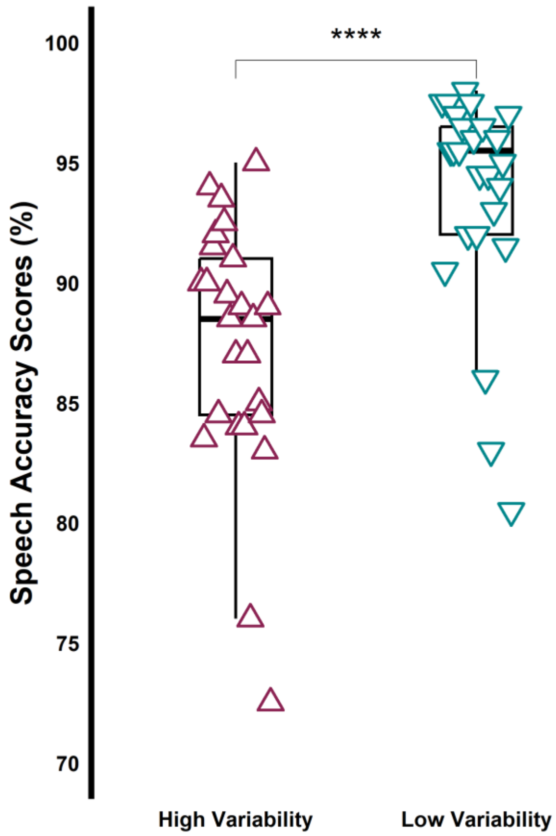
Abstract

Background. Variability of talker voice can negatively affect speech perception. This multitalker processing cost may arise due to use of cognitive resources during speech processing. Previous research on talker variability has primarily used single words/digits; however, in the real-world, listeners must understand speech in whole sentences. Thus, the purpose of the study was to examine whether multitalker processing cost persists in ecologically valid condition (with sentences), and to study the cognitive factors that may mediate speech perception performance in individuals with normal hearing.

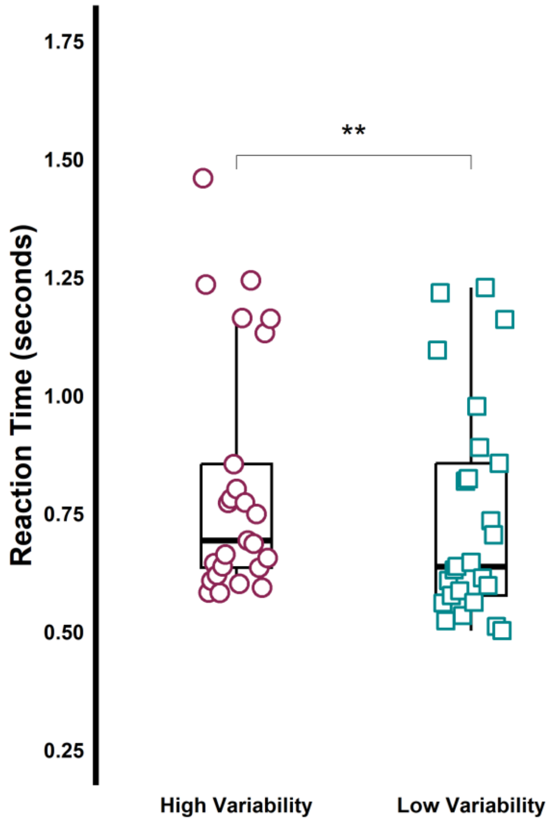
Methods. Twenty-five adults with normal hearing underwent speech-in-noise testing in single talker (low variability) and multiple different talkers in succession (high variability) conditions. To measure reaction time, the task was to indicate the first word of the sentence heard as fast as possible, and participants then repeated the entire sentence to measure accuracy. Additionally, cognition was assessed using the NIH Toolbox Cognition Battery, and correlations between these scores and speech perception accuracy and reaction times with high and low-variability sentences were examined.

Results. Speech recognition accuracy and speed were both significantly lower for the high compared to low variability conditions. Additionally, the cognitive domains of crystallized cognition showed a significant moderate positive correlation with speech accuracy and fluid cognition showed a significant moderate negative correlation with reaction time, for both talker conditions.

Conclusions. These findings suggest that even in individuals with normal hearing, adapting to a constantly changing target talker significantly hinders sentence perception. Testing in the clinic with stimuli with greater variability is expected to be more consistent with individuals' difficulties surrounding speech communication in daily life.



****
100
95
90
85
80
75
70
Speech Accuracy Scores (%)
High Variability
Low Variability



**
1.75
1.50
1.25
1.00
0.75
0.50
0.25
Reaction Time (seconds)
High Variability
Low Variability

Speech Perception in Noise in Individuals with Misophonia: Evidence from The Audible Contrast Threshold Test

Hamssika Sudhakar (All India Institute of Speech and Hearing)*; Prashanth Prabhu (All India Institute of Speech and Hearing)

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Podium

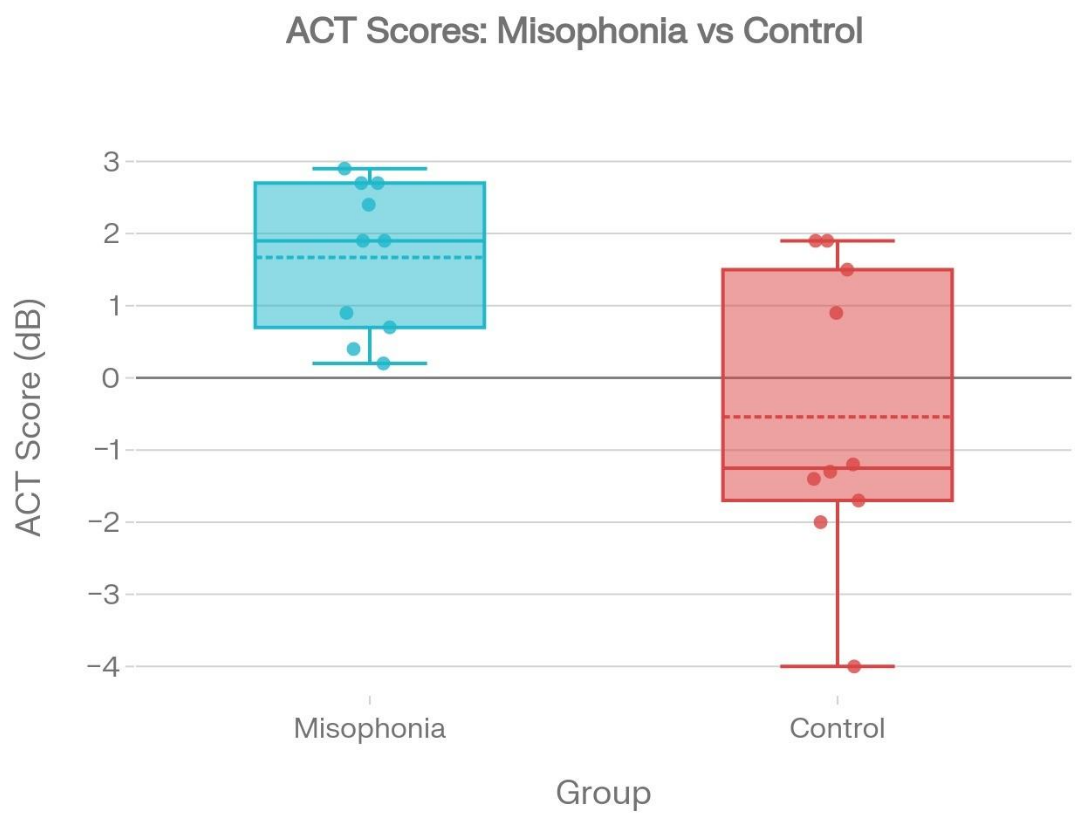
Abstract

Background. Misophonia is characterized by an abnormally strong emotional or physiological response to specific sounds, with evidence suggesting that emotional reactivity adversely affects auditory attention and speech perception in noise in affected individuals (Ozdes et al., 2025; Kim et al., 2023). The Audible Contrast Threshold (ACT) test is a subjective tool designed to quantify speech understanding in noisy environments. The present study aimed to investigate speech perception in noise using the ACT test in individuals with misophonia.

Methods. Twenty participants were recruited and divided equally into two groups: a misophonia group (n = 10) and a control group (n = 10). Individuals with misophonia were identified using Schröder's diagnostic criteria (2014), and severity was assessed using the Revised Amsterdam Misophonia Scale (AMISOS-R). Pure-tone audiometry and ACT were administered using the MedRx software.

Results. Both groups demonstrated normal hearing sensitivity on pure-tone audiometry, ruling out peripheral hearing loss as a confounding factor. Data for both groups satisfied the assumption of normality (Shapiro-Wilk test, $p > 0.05$). An independent-samples t-test revealed a statistically significant difference in ACT scores between the misophonia group (Mean = 1.67 dB, SD = 1.03) and the control group (Mean = −0.54 dB, SD = 1.98); $t(18) = 3.13$, $p = 0.006$. A large effect size was observed (Cohen's $d = 1.40$), indicating that individuals with misophonia required significantly higher contrast to understand speech in noise compared to controls.

Conclusions. The elevated ACT scores in the misophonia group suggest that heightened emotional reactivity to sounds may interfere with the central auditory processing of speech in degraded listening conditions. These results highlight the importance of incorporating speech-in-noise assessments in the audiological evaluation of individuals with misophonia to better characterize their auditory profile.


ACT Scores: Misophonia vs Control
ACT Score (dB)
3
2
1
0
−1
−2
−3
−4
Misophonia
Control
Group

A Comparison of Speech Recognition Between Humans and Modern Machine Systems

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Podium

Abstract

Background. Models that can accurately predict human speech recognition could have many uses. Recent machine speech recognition systems have been shown to explain aspects of human brain responses to speech. However, it is unclear how well these systems account for human speech recognition behavior, as they have not been extensively compared to humans on speech recognition tasks.

Methods. We compared the performance of humans to 5 recent industry speech recognition systems (Parakeet-TDT-0.6B-v2, SeamlessM4T-v2, Whisper-large-v3, Whisper-large-v1, Whisper-tiny-v1) on the task of reporting the words of a 2-second speech excerpt to which a distortion had been applied. Humans and models each transcribed all the words in each excerpt. To avoid the possibility that models might be better at reporting low-frequency words, scoring was restricted to a set of 230 common words, one of which was present per excerpt. Motivated by the idea that an accurate model should be able to reproduce human performance across a wide range of conditions, we measured human and model performance across 37 types of distortions, most of which were varied parametrically.

Results. The models exhibited high correlations with the pattern of human performance across distortions (Fig. 1). However, some industry models exhibited superhuman performance, showing substantially higher intelligibility than humans for most types of distortion. Whisper-tiny-v1, which was trained on the same data as Whisper-large-v1 (around 78 years of continuous speech) but had a smaller architecture, underperformed humans, suggesting that superhuman performance may be restricted to large models (Fig. 2).

Conclusions. Current industry speech recognition models exhibit human-like patterns of performance across speech distortions, but some such systems now exhibit superhuman performance. The results motivate scientific efforts to build models whose architectures and training data enable them to match human levels of performance.

Figure 1. Human vs. model performance on all conditions (N=37)

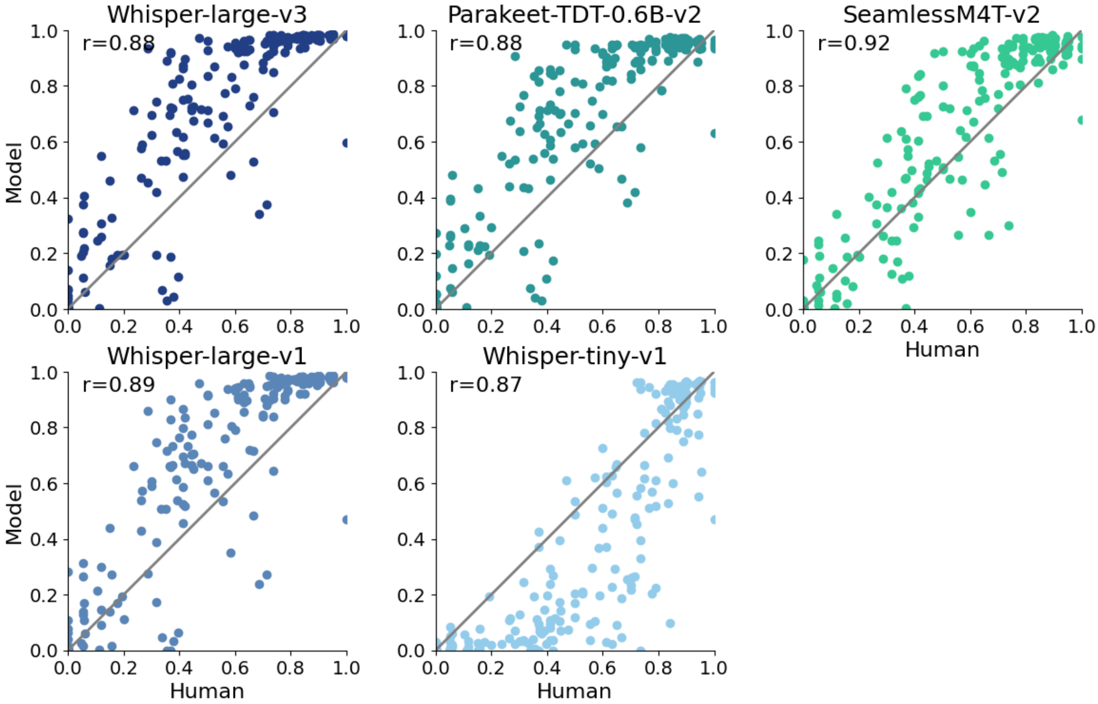
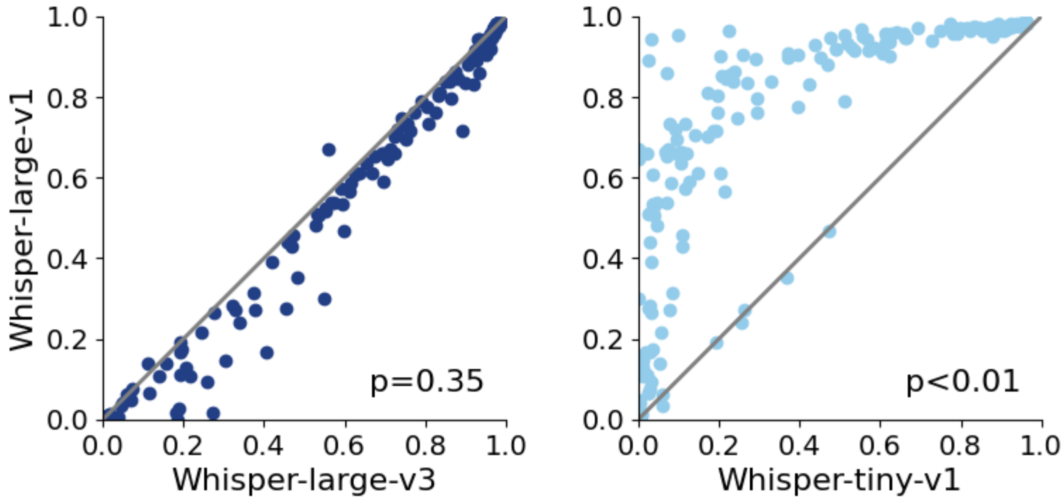


Figure 2. Model comparisons with Whisper-large-v1



Test–Retest Reliability of the Audible Contrast Threshold (ACT) in Young Normal-Hearing Adults

Kavya Sreekumar (AIISH, Mysuru)*; Prashanth Prabhu (AIISH, Mysuru)

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Poster

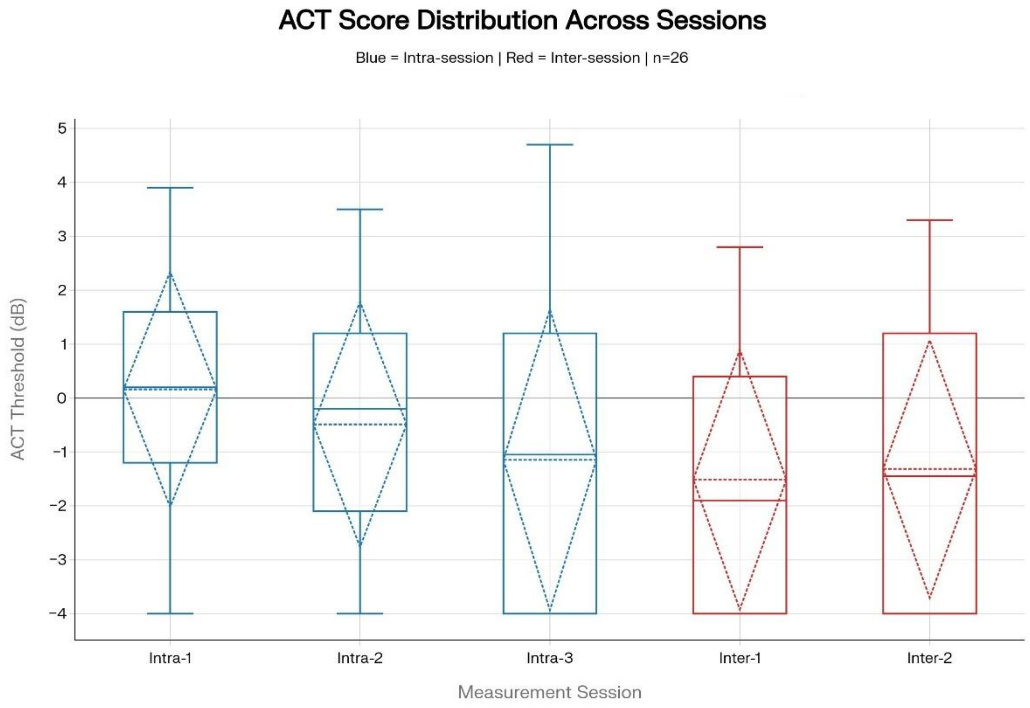
Abstract

Background. The Audible Contrast Threshold (ACT) is a clinically efficient, language-independent tool for predicting aided speech-in-noise performance. ACT captures spectro-temporal modulation sensitivity, providing a suprathreshold index of auditory processing that correlates with real-world listening difficulty. Limited evidence exists on its test–retest reliability, which is essential for validating ACT for longitudinal monitoring and clinical research.

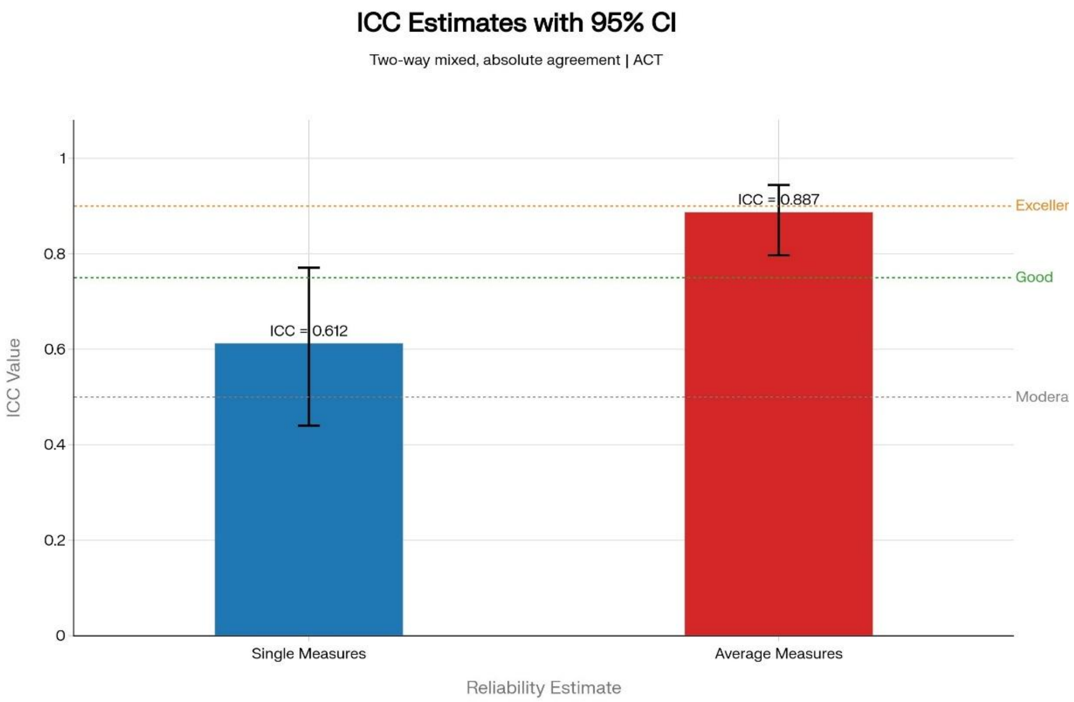
Methods. Thirty normal-hearing adults (18–25 years) completed repeated ACT measurements across two sessions one week apart. Each participant completed three intra-session and two inter-session recordings (five observations total). Test–retest reliability was evaluated using two-way mixed-effects ICC, Cronbach's alpha, and Bland-Altman analysis.

Results. ICC revealed moderate single-measure reliability (ICC = 0.612; 95% CI: 0.440–0.771) and good-to-excellent average-measure reliability (ICC = 0.887; 95% CI: 0.797–0.944), with significant consistency across measurements ($F(25,100) = 10.343$, $p < 0.001$). Cronbach's alpha showed excellent internal consistency ($\alpha = 0.903$), with item-total correlations of 0.726–0.794. Bland-Altman analysis revealed minimal inter-session bias (−0.20 dB; 95% LoA: −3.27 to +2.87 dB), confirming negligible systematic error. Intra-session pairs showed slightly wider limits of agreement (bias: +0.65 to +1.31 dB), reflecting a mild practice effect.

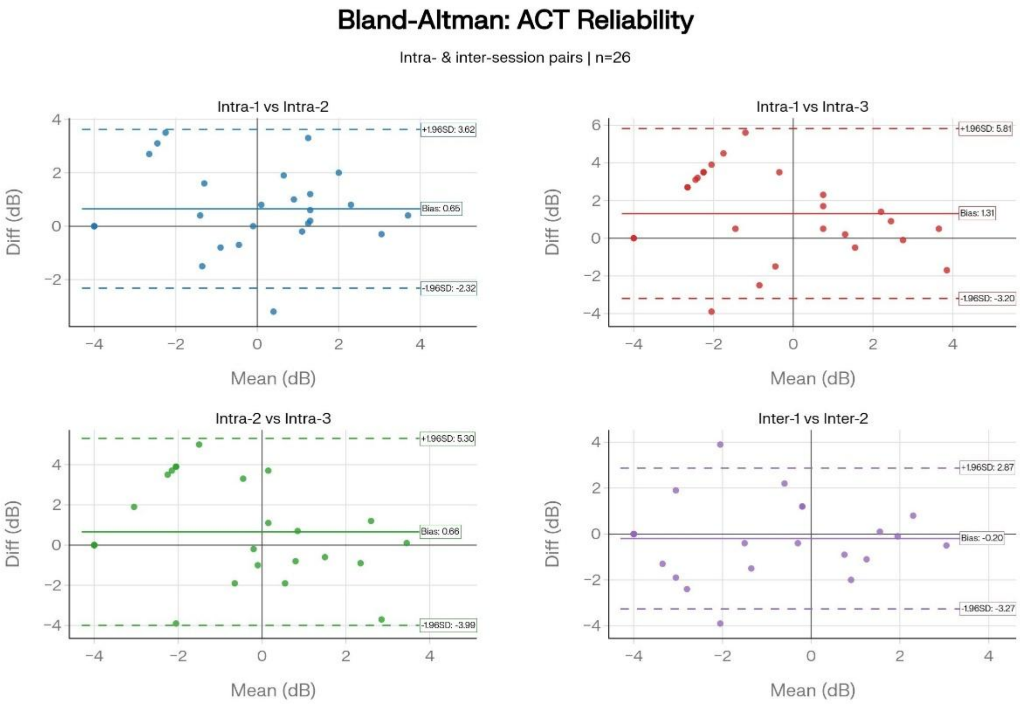
Conclusions. ACT demonstrates moderate-to-good test–retest reliability and excellent internal consistency, with stronger reliability when thresholds are averaged. Near-zero inter-session bias supports ACT's clinical stability over time. Clinicians should average repeated ACT recordings for optimal stability. This validates ACT as a reliable, language-independent measure for supra-threshold processing. Further research is needed across diverse populations, including older adults and individuals with sensorineural hearing loss.



ACT Score Distribution Across Sessions
Blue = Intra-session | Red = Inter-session | n=26
ACT Threshold (dB)
5
4
3
2
1
0
−1
−2
−3
−4
Intra-1
Intra-2
Intra-3
Inter-1
Inter-2
Measurement Session



ICC Estimates with 95% CI
Two-way mixed, absolute agreement | ACT
ICC Value
1
0.8
0.6
0.4
0.2
0
ICC = 0.887
ICC = 0.612
Exceller
Good
Modera
Single Measures
Average Measures
Reliability Estimate


Bland-Altman: ACT Reliability
Intra- & inter-session pairs | n=26
Intra-1 vs Intra-2
+1.96SD: 3.62
Bias: 0.65
-1.96SD: -2.32
Intra-1 vs Intra-3
+1.96SD: 5.81
Bias: 1.31
-1.96SD: -3.20
Intra-2 vs Intra-3
+1.96SD: 5.30
Bias: 0.66
-1.96SD: -3.99
Inter-1 vs Inter-2
+1.96SD: 2.87
Bias: -0.20
-1.96SD: -3.27
Diff (dB)
Mean (dB)

Effect of Online Music-Based Workshops on Music Perception in Older Adults with Hearing Loss

Alexis Whittom (Université Laval)*; Alex Bégin (Université Laval); Sarah Haloui (Université Laval); Angelina Lynne (Université Laval); Marjorie Fiset (Université Laval); Valerie Peters (Université Laval); Pascale Tremblay (Université Laval); Isabelle Blanchette (Université Laval); Andréanne Sharp (Université Laval)

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Poster

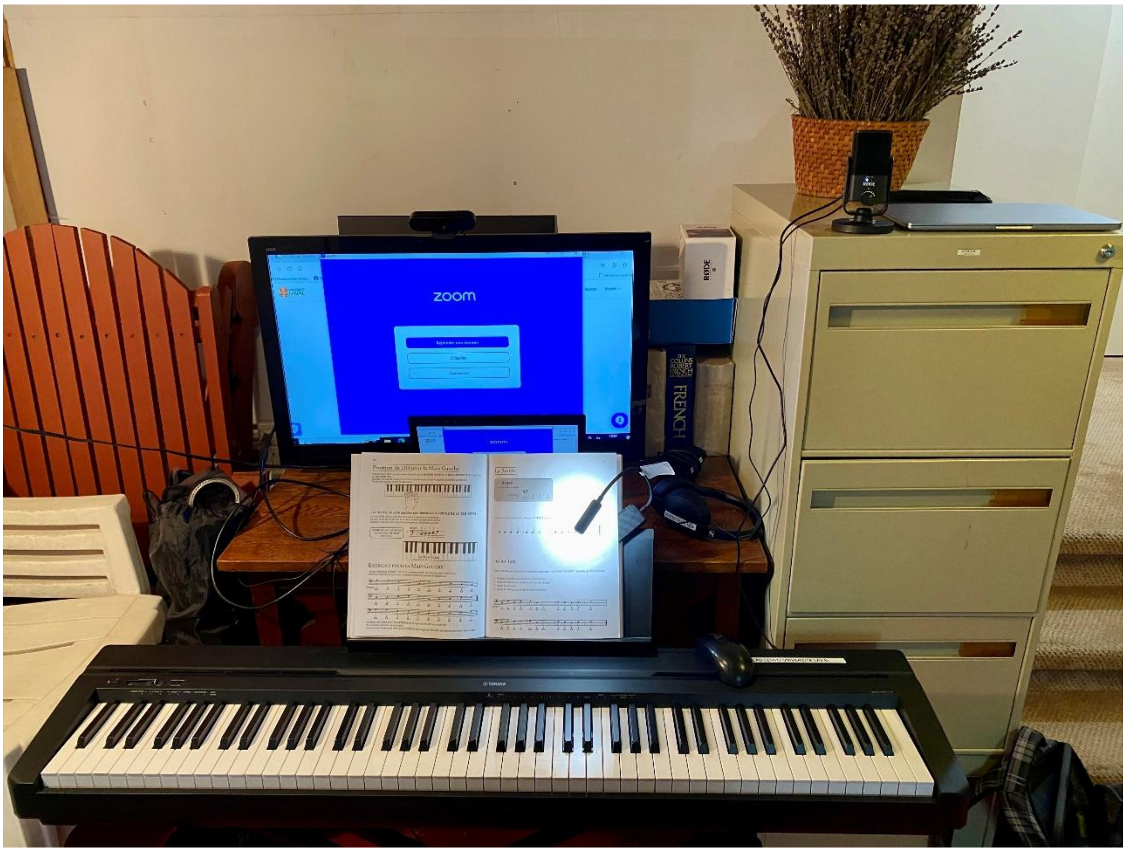
Abstract

Background. Around 30% of individuals aged 60 and over experience some degree of hearing loss, a condition often linked to reduced engagement in leisure activities. Hearing aids are widely used to compensate for auditory impairment, yet many users continue to report dissatisfaction with sound quality when listening to music. Lifelong engagement in musical activities may have a protective effect on auditory abilities, suggesting that musical education could be a promising tool for hearing rehabilitation. The main objective of this project was to develop online music-based activities for older adults with hearing loss and examine their impact on music perception.

Methods. Twenty-seven older adults (60–89 years) with a mean PTA at 3–4–6–8 kHz of 67±12 dB HL were assigned to piano learning (G1; n=8), music listening (G2; n=11), or discussion activities (G3; n=8). Weekly one-hour sessions included basic piano lessons (G1), guided listening (G2), or structured discussions on predetermined topics (G3). Participants completed a musical perception assessment before (T1) and after (T2) the 12-week program. A shortened version of the Montreal Battery of Evaluation of Amusia (MBEA) assessed scale, contour, and rhythm processing, while an instrument-discrimination task evaluated timbre perception. Nine normal-hearing adults completed the same tasks at both time points without training, confirming the absence of a practice effect.

Results. Since no significant group differences were observed at either time point, data from all three groups were combined. Significant improvements emerged between T1 and T2 for the MBEA score ($p < .001$; very large effect size = 0.801) and for instrument discrimination ($p = .017$; large effect size = 0.597).

Conclusions. These preliminary findings suggest that all proposed activities—musical or not—may enhance melody, rhythm, and timbre recognition in older adults with hearing loss.


zoom
FRENCH

Extended High-Frequency (EHF) Hearing and Speech-in-Noise Difficulty Despite Normal Conventional High-Frequency Hearing and Preserved Word Recognition in Quiet

Navid Shahnaz (University of British Columbia)*; Ragul Loganathan (University of British Columbia); Mohammadreza Rahmani Manesh (University of British Columbia); Eytan A. David (University of British Columbia); Valter Ciocca (University of British Columbia)

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Poster

Abstract

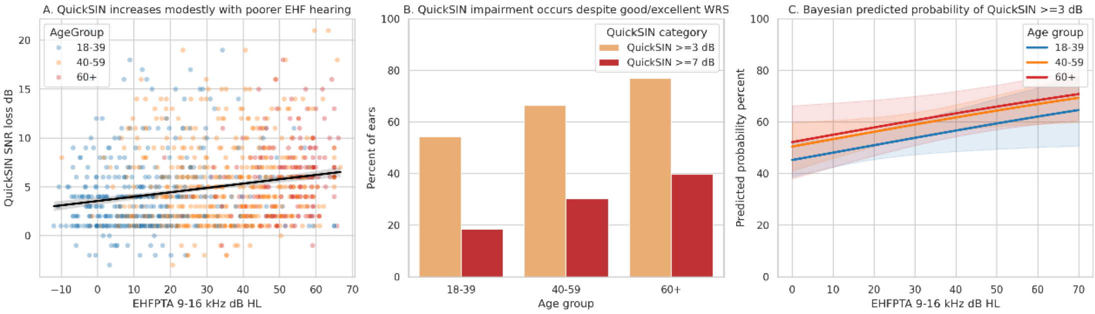
Background. Conventional audiometry and word recognition in quiet may fail to explain difficulty understanding speech in noise. Extended high-frequency (EHF; 9–16 kHz) hearing may capture auditory dysfunction not reflected by conventional high-frequency pure-tone average (HFPTA; 1, 2, 3, and 4 kHz) or word recognition scores (WRS).

Objective: To determine whether EHF pure-tone average (EHFPTA) predicts QuickSIN performance in ears with normal HFPTA and preserved WRS.

Methods. This retrospective analysis included 1,095 ears from 593 adults aged 18–82 years (mean 45.0 ± 13.4 years) with normal HFPTA (≤25 dB HL), excellent NU-6 WRS in quiet (≥88%), EHF thresholds, and QuickSIN results. Age was modeled categorically: 18–39 years (407 ears), 40–59 years (520 ears), and ≥60 years (168 ears). EHFPTA was the average threshold from 9 to 16 kHz. QuickSIN was modeled as a metric outcome and categorically using ≥3 dB and ≥7 dB SNR loss cut points. Bayesian regression tested EHFPTA after adjustment for HFPTA, WRS, and age.

Results. Despite excellent WRS, 62.8% of ears had QuickSIN SNR loss ≥ 3 dB and 25.7% had SNR loss ≥ 7 dB. Among ears with good WRS, 91.2% had SNR loss ≥3 dB and 76.5% had SNR loss ≥7 dB. In models restricted to ears with normal HFPTA and WRS ≥76%, each 10 dB worsening in EHFPTA increased the odds of QuickSIN SNR loss ≥3 dB (OR = 1.12, 95% credible interval [1.01, 1.24]). The association with SNR loss ≥7 dB was weaker (OR = 1.05, 95% credible interval [0.94, 1.17]). As a metric outcome, EHFPTA showed a modest positive association with QuickSIN SNR loss (β = 0.18 dB per 10 dB worsening), although the credible interval included zero. Adding EHFPTA modestly improved prediction beyond HFPTA, WRS, and age.

Conclusions. Speech-in-noise deficits were common despite normal HFPTA and preserved WRS. EHFPTA helped identify mild QuickSIN SNR loss, supporting EHF audiometry as an adjunct when listening difficulty is not captured by conventional audiometry or WRS alone.



A. QuickSIN increases modestly with poorer EHF hearing
AgeGroup
18-39
40-59
60+
QuickSIN SNR loss dB
EHFPTA 9-16 kHz dB HL
B. QuickSIN impairment occurs despite good/excellent WRS
QuickSIN category
QuickSIN >=3 dB
QuickSIN >=7 dB
Percent of ears
Age group
C. Bayesian predicted probability of QuickSIN >=3 dB
Age group
18-39
40-59
60+
Predicted probability percent
EHFPTA 9-16 kHz dB HL

Cadenza: Lyric Intelligibility Prediction for Musical Enjoyment

Rebecca Vos (University of Salford)*; Simone Graetzer (University of Salford); Gerardo Roa Dabike (University of Sheffield); Michael Akeroyd (University of Nottingham); Scott Bannister (University of Leeds at the time of this work, currently at University of Manchester); Jon Barker (University of Sheffield); Trevor Cox (University of Salford); Bruno Fazenda (University of Salford); Jennifer Firth (University of Nottingham); Alinka Greasley (University of Leeds); William Whitmer (University of Nottingham)

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Poster

Abstract

Background. Listening to music has health and well-being benefits, and understanding lyrics is part of music enjoyment. However, hearing loss and hearing aids can reduce listeners' ability to understand lyrics. This is a significant concern; more than 430 million people worldwide have disabling hearing loss. Improving and predicting lyric intelligibility is not trivial, and can be affected by vocal style, song genre, mixing production and listener hearing ability. The performance of speech intelligibility prediction metrics for sung lyrics is unreliable. In the Cadenza Lyric Intelligibility Prediction challenge (CLIP1), entrants were asked to improve on our baseline prediction models for different levels of hearing loss.

Methods. The task was to predict lyric intelligibility from a musical excerpt containing 5-10 words. A dataset of 11,072 listener-labelled samples was provided, with two baseline systems; a non-intrusive use of Short-Term Objective Intelligibility (STOI) with an audio source separation model (HDemucs), and a Whisper-based Automatic Speech Recognition (ASR) system (see figure).

Results. 22 teams submitted 27 systems, of which 16 outperformed the best (Whisper-based) baseline;

- 24 systems used foundation models for high-level acoustic representation (including Whisper).

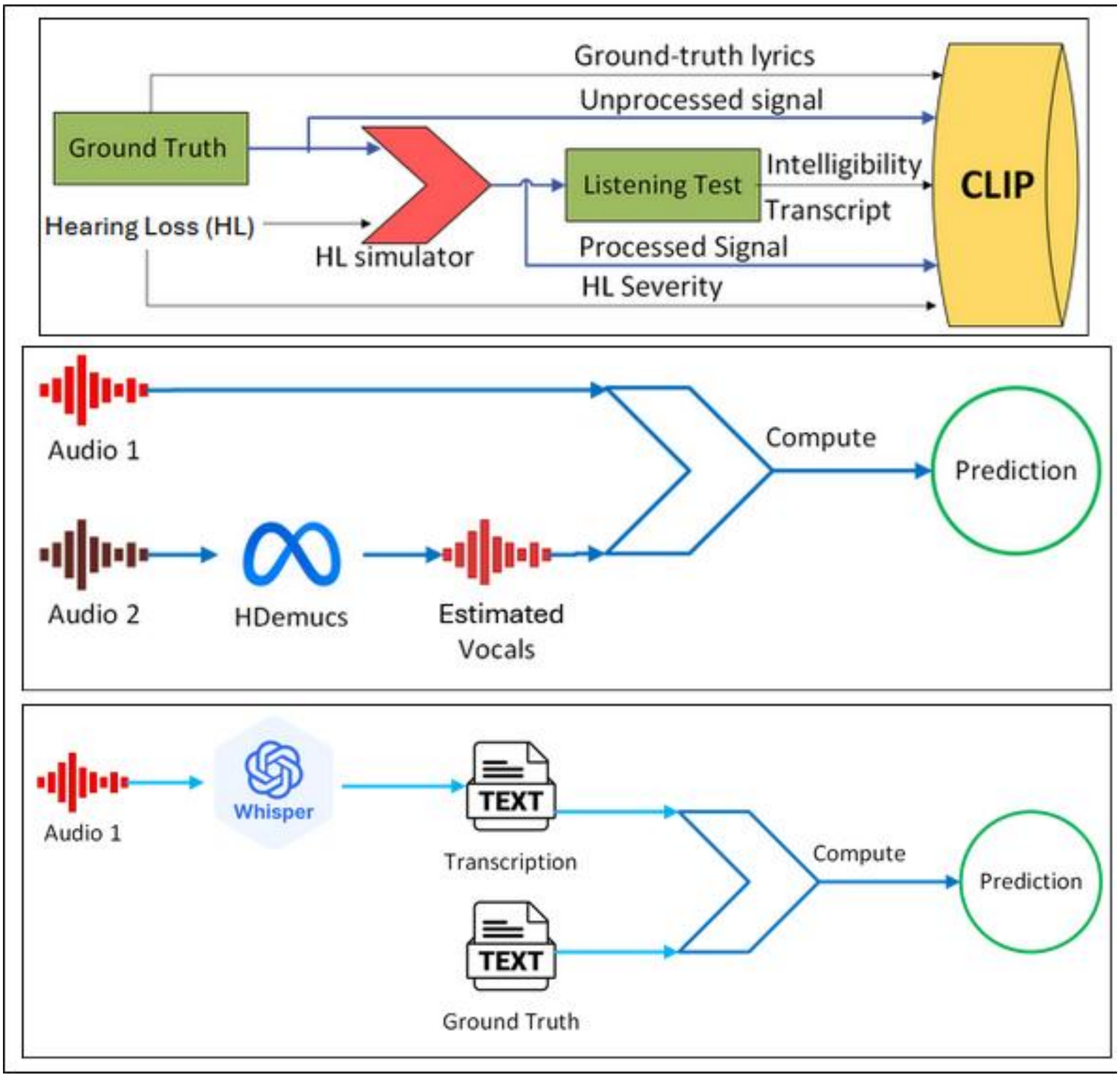
- 4 systems used text encoders including Hugging Face T5 models.

- Additional features were used, including mel-frequency cepstral coefficients (MFCC), signal-to-noise ratio (SNR) and pitch.

- 4 systems used data augmentation methods.

Conclusions. The high errors indicate opportunities for improvement. Sources of variability in the data included singing style and accompaniment, which could be better modelled by larger, more varied datasets. We announce the second Cadenza Lyric Intelligibility Prediction challenge (CLIP2). Datasets will feature thousands of excerpts

from AI-Generated songs, simulating different listening scenarios and genres. Launch: 1/6/26, Submission: 1/11/26.



Effects of stimulus degradation and attention on hierarchical auditory processing: Evidence from brainstem and cortical responses

Srikanth Nayak (Yenepoya Medical College, Yenepoya University)*; Arivudai Nambi (All India Institute of Speech and Hearing); Jayashree Bhat (Yenepoya university)

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Poster

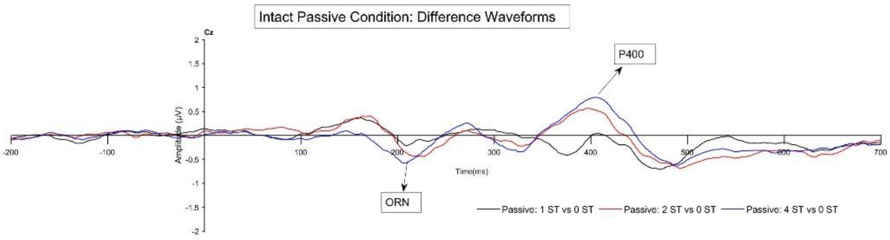
Abstract

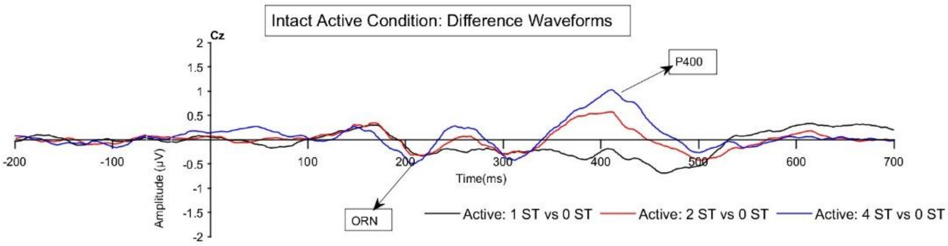
Introduction. Understanding speech in the presence of noise requires efficient neural encoding throughout the auditory pathway. While the role of the brainstem to the cortex is well understood, their interaction under attention is less explored. Further, degraded stimuli may require compensatory neural mechanisms for perception. The current study investigates subcortical and cortical neural coding and interactions for intact and degraded stimuli.

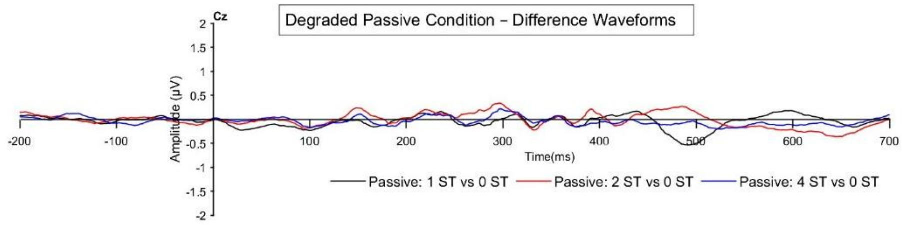
Method. Simultaneous Brainstem and cortical responses were recorded in 43 young adults with normal hearing. Nonsense CVVC stimuli were pitch-shifted to create four semitone conditions (0, 1, 2, 4 ST). For degraded stimuli, temporal fine structure was removed. Responses were recorded under passive and active listening conditions. Brainstem processing was assessed using the frequency-following response, and cortical processing was assessed using the ORN and P400. Linear mixed-effects models evaluated effects of Stimulus, Task, and pitch differences.

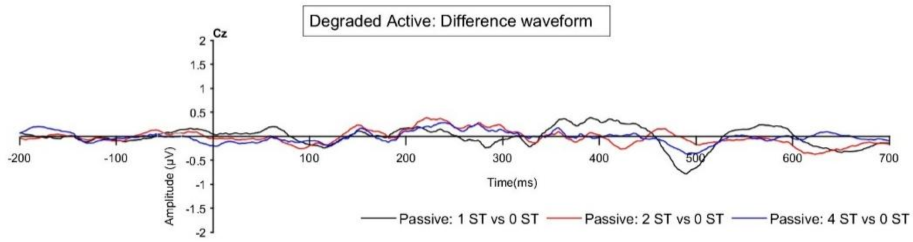
Results. FFR amplitude showed significant main effects of Stimulus, Task, and Semitone, with a significant Stimulus × Semitone interaction ($p < .001$). Intact stimuli elicited higher amplitude and greater sensitivity to pitch changes, whereas degraded stimuli showed reduced encoding. ORN amplitude demonstrated a significant effect of Stimulus ($p = .013$), with more negative responses to intact stimuli and no reliable ORN observed for degraded stimuli. P400 amplitude revealed significant main effects of Stimulus ($p = .006$) and Difference Waveform ($p < .001$), and a Stimulus × Difference Waveform interaction ($p < .001$). P400 varied across pitch differences for intact stimuli ($p < .001$) but not for degraded stimuli ($p = .254$).

Conclusion. Degradation disrupts pitch processing across the auditory pathway. Brainstem encoding is reduced, early cortical responses depend on intact signals, and late cortical stages fail to differentiate pitch, underlining the importance of spectral cues.


Intact Passive Condition: Difference Waveforms
Cz
Amplitude (μV)
Time(ms)
P400
ORN
Passive: 1 ST vs 0 ST
Passive: 2 ST vs 0 ST
Passive: 4 ST vs 0 ST


Intact Active Condition: Difference Waveforms
Cz
Amplitude (μV)
Time(ms)
P400
ORN
Active: 1 ST vs 0 ST
Active: 2 ST vs 0 ST
Active: 4 ST vs 0 ST


Degraded Passive Condition – Difference Waveforms
Cz
Amplitude (μV)
Time(ms)
Passive: 1 ST vs 0 ST
Passive: 2 ST vs 0 ST
Passive: 4 ST vs 0 ST


Degraded Active: Difference waveform
Cz
Amplitude (μV)
Time(ms)
Passive: 1 ST vs 0 ST
Passive: 2 ST vs 0 ST
Passive: 4 ST vs 0 ST

Cortical speech tracking in older adults with hearing loss

Fauve Duquette-Laplante (Flinders University); Varghese Peter (University of Sunshine Coast); Christian Boyle (Flinders University); Ajay Raman (University of Queensland); Shivani Prabhu (Flinders University); Mridula Sharma (Flinders University)*

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Poster

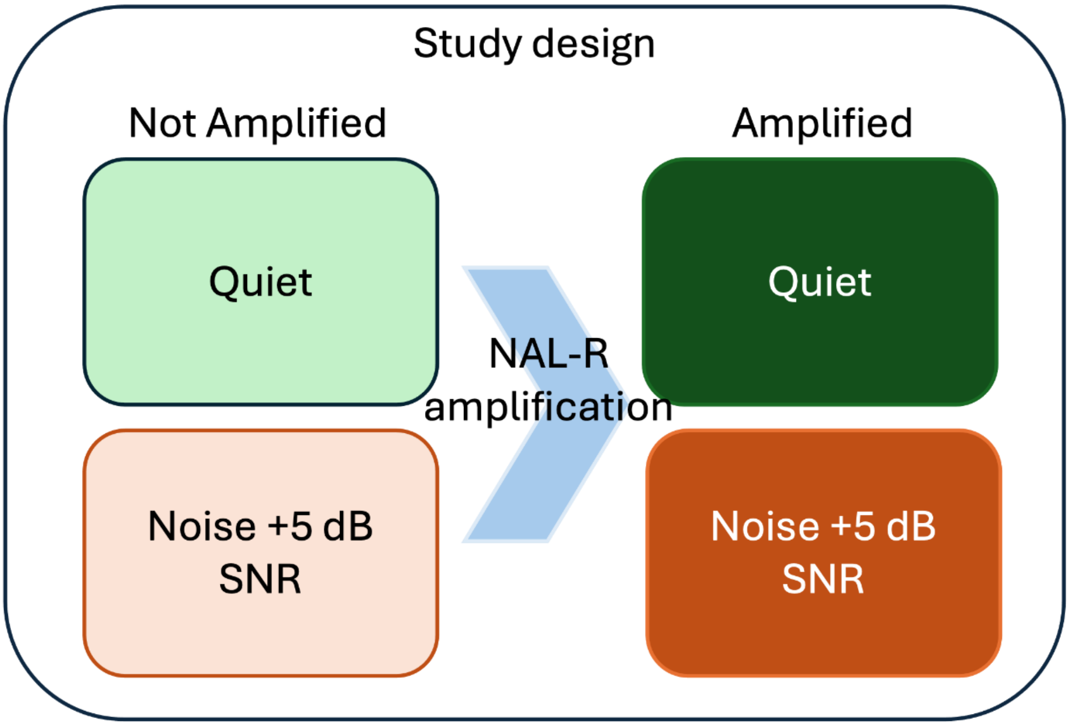
Abstract

In older adults, the most common type of hearing loss is bilateral sensorineural that impacts audibility as well as auditory processing. Consequently, hearing aids that address audibility deficits provide limited benefit to older adults in noisy settings. In this study, we aim to investigate the effects of mild to moderate hearing loss on listening in noise in older adults.

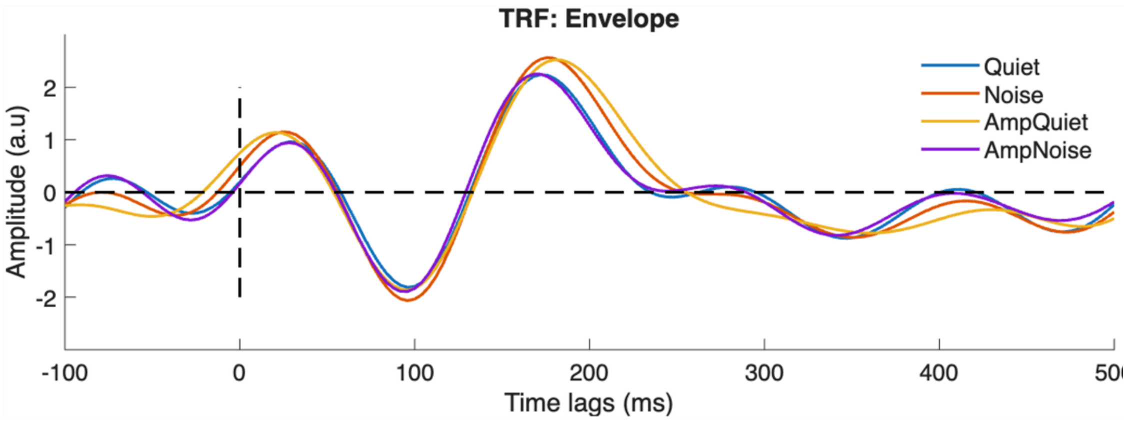
We used a temporal response framework (TRF) to measure cortical speech tracking to amplified and unamplified conversations in quiet and in noise in 13 older adults, 74 years (stddev 3.8). We measured cortical speech tracking to 15-minute podcasts presented in free field at 1.3 meters at 65dB SPL (±2dB) in quiet and then in presence of 8-talker babble at 5dB SNR. We also measured their speech reception threshold (SRT) on ToLD-U unamplified and amplified. Ten adults did not wear hearing aids while 2 had hearing aids but used sparingly and only one was a regular hearing aid user. For consistency, all participants were tested without devices and for amplified paradigm, podcasts (with and without noise) were amplified using the formula NAL-R.

Results showed that amplification significantly improved the speech reception threshold when tested behaviourally [$F(1, 12) = 44$, <0.001] but did not improve the predictive accuracy, which was similar in quiet and in noise, with and without amplification. Contrary to the hypothesis, hearing loss and noise did not impact the predictive accuracy of speech features in this population.

The present study shows that older adults with hearing loss can listen to speech in presence background noise. There is high level of individual variability, however. Perhaps the participants have adequate compensatory strategies such as language skills, low frequency hearing and selective attention. We are still collecting data and aim to include older adults without hearing loss to determine if the predictive accuracy is better especially in noise in absence of a hearing loss.



Study design
Not Amplified
Amplified
Quiet
NAL-R
amplification
Quiet
Noise +5 dB
SNR
Noise +5 dB
SNR



TRF: Envelope
Quiet
Noise
AmpQuiet
AmpNoise
Amplitude (a.u)
Time lags (ms)

Session 2.B. Cognition, Listening Effort & Brain Connectivity

Neuro-cognitive models, EEG/fMRI correlates, and individual differences in listening effort.

Chaired by Dr. Yue Zhang.

Decoding of speech acoustics from EEG: going beyond the amplitude envelope

Alexis MacIntyre (University of Cambridge)*; Clément Gaultier (Institut Pasteur); Tobias Goehring (University of Zurich, University of Cambridge)

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Featured talk

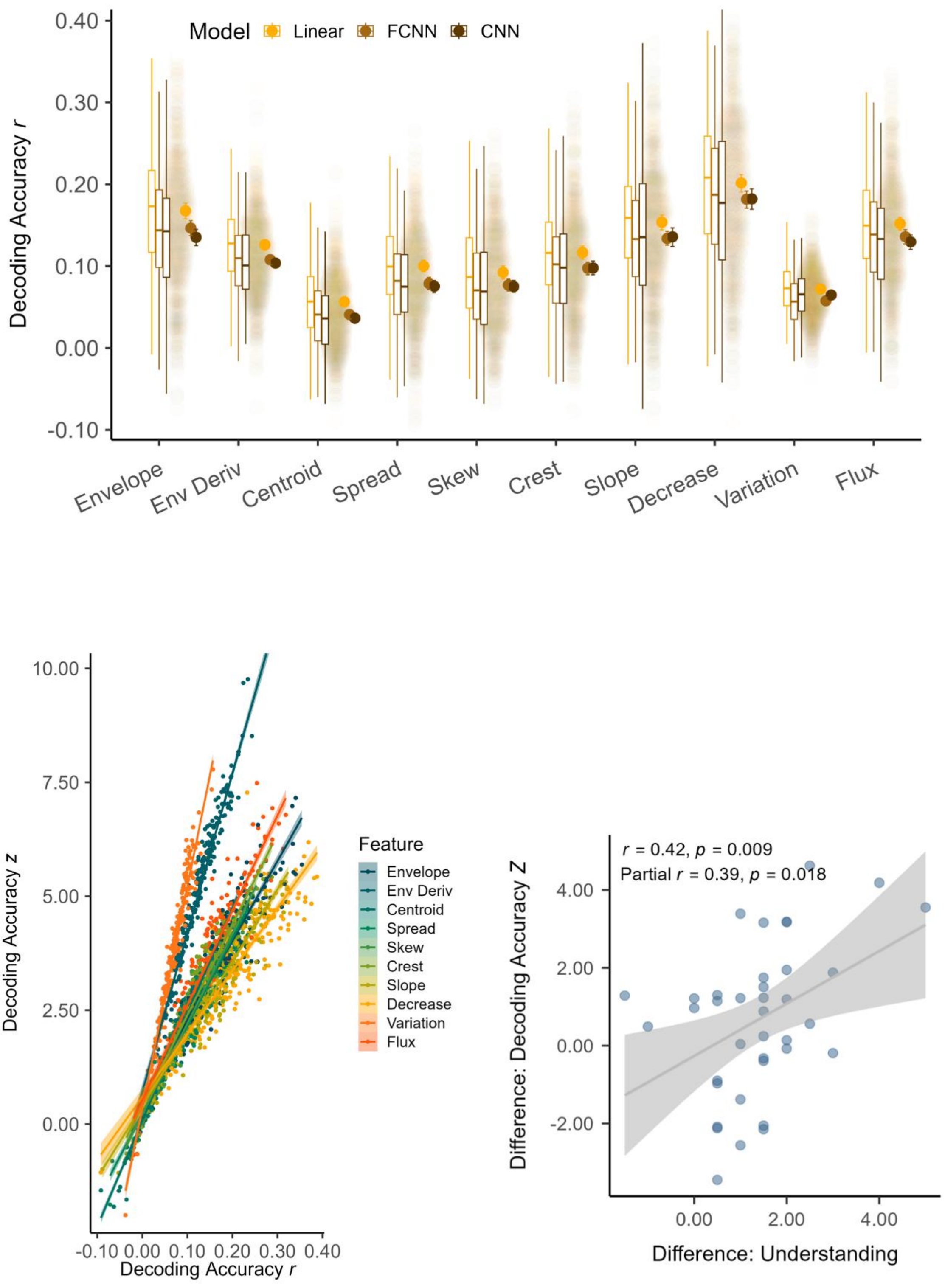
Abstract

Background. During speech perception, properties of the acoustic stimulus can be reconstructed from the listener's brain using electroencephalography (EEG). Most studies employ the amplitude envelope as a decoding target; however, speech acoustics can be characterised on multiple dimensions, including as spectral descriptors. The current study assesses how robustly an extended acoustic feature set can be decoded from EEG under varying levels of intelligibility and acoustic clarity.

Methods. We recorded EEG from 38 young adults who heard intelligible and non-intelligible speech that was either unprocessed or spectrally degraded via vocoding. We extracted envelope-based features, as well as descriptors characterising instantaneous properties of the spectrum (e.g. spectral slope) or spectral change over time (e.g. spectral flux). Robustness of feature decoding was established using linear and nonlinear model architectures and, for linear decoders, by standardising accuracy (Pearson's r) using randomly permuted surrogate data.

Results. Linear and nonlinear models perform similarly well and produce a consistent pattern of results across features and conditions. Converting r to Z-scores scaled by random data revealed noise floor differences between features. Decoding accuracy varied by spectral degradation and intelligibility for some features, but such differences were reduced in more robustly decoded features, suggesting reconstruction is mainly driven by generalised auditory processing. Spectral flux uniquely predicts individual comprehension after controlling for acoustics.

Conclusions. Linear decoders perform comparably to nonlinear decoders in capturing the EEG response to speech acoustics beyond the amplitude envelope, with the reconstructive accuracy of some features reflecting understanding and spectral clarity. This sheds light on how sound properties are differentially represented by the brain and shows potential for clinical applications moving forward.


Model
Linear
FCNN
CNN
Decoding Accuracy r
0.40
0.30
0.20
0.10
0.00
-0.10
Envelope
Env Deriv
Centroid
Spread
Skew
Crest
Slope
Decrease
Variation
Flux
Decoding Accuracy z
10.00
7.50
5.00
2.50
0.00
-0.10 0.00 0.10 0.20 0.30 0.40
Decoding Accuracy r
Feature
Envelope
Env Deriv
Centroid
Spread
Skew
Crest
Slope
Decrease
Variation
Flux
r = 0.42, p = 0.009
Partial r = 0.39, p = 0.018
Difference: Decoding Accuracy Z
4.00
2.00
0.00
-2.00
0.00
2.00
4.00
Difference: Understanding

Auditory Processing Differences in Stuttering: Evidence from Load Theory of Attention

Fjorda Kazazi (UCL)*

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Podium

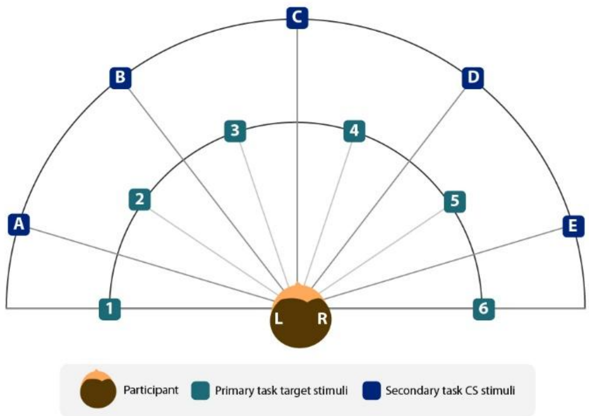
Abstract

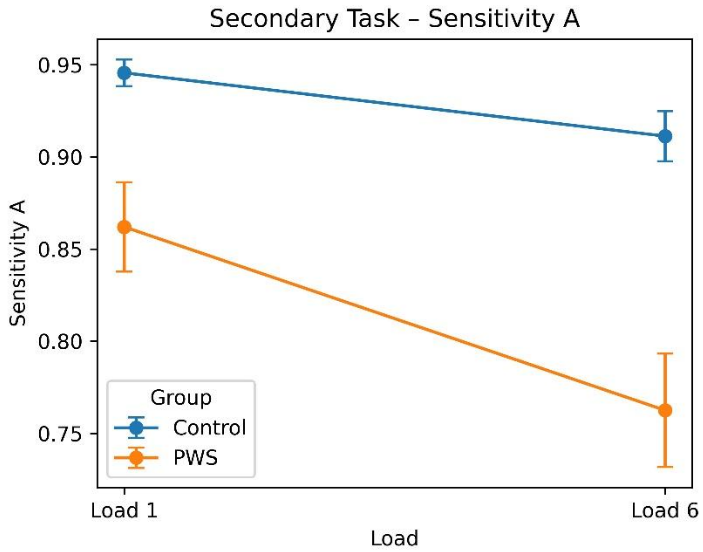
Background. The load theory of attention proposes that the detection of irrelevant information depends on perceptual load, with higher load reducing processing of distractors. This framework is relevant to listening effort and auditory scene analysis, where cognitive resources must be allocated across competing sounds. People who stutter (PWS) may rely more on cognitive resources during speech, but it remains unclear whether similar differences extend to non-speech spatial auditory processing.

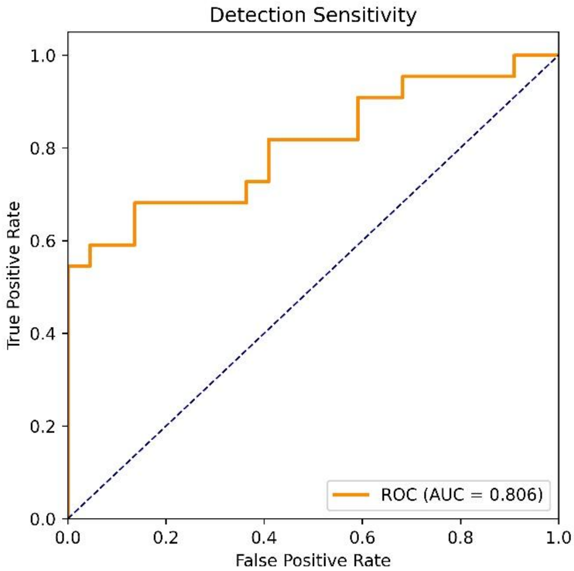
Methods. Twenty-two PWS and 22 matched controls completed a dual-task spatial auditory paradigm with varying perceptual load. Auditory stimuli were spatialised to simulate real-world listening environments. Primary task performance (accuracy and reaction time) and secondary task detection sensitivity (DS) were measured to assess auditory processing under cognitive demand.

Results. Both groups showed reduced accuracy and increased reaction times with higher load. However, PWS showed lower detection sensitivity overall, with a significant Group × Load interaction indicating greater impairment under high load. Exploratory analysis showed that DS predicted group membership (AUC = 0.81).

Conclusions. These findings suggest reduced auditory processing efficiency in PWS under high cognitive load in spatial listening conditions. The results extend load theory to auditory cognition in stuttering and highlight detection sensitivity as a potential behavioural marker of listening effort in complex acoustic environments.




Secondary Task – Sensitivity A
Sensitivity A
0.95
0.90
0.85
0.80
0.75
Group
Control
PWS
Load 1
Load 6
Load


Detection Sensitivity
True Positive Rate
1.0
0.8
0.6
0.4
0.2
0.0
ROC (AUC = 0.806)
0.0
0.2
0.4
0.6
0.8
1.0
False Positive Rate

Effect of Reverberation and Signal to Noise Ratio on Listening Effort in Native and Non-native Kannada Speakers

Shivani Sharma (AIISH)*; Kavassery Venkateswaran Nisha (AIISH)

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Podium

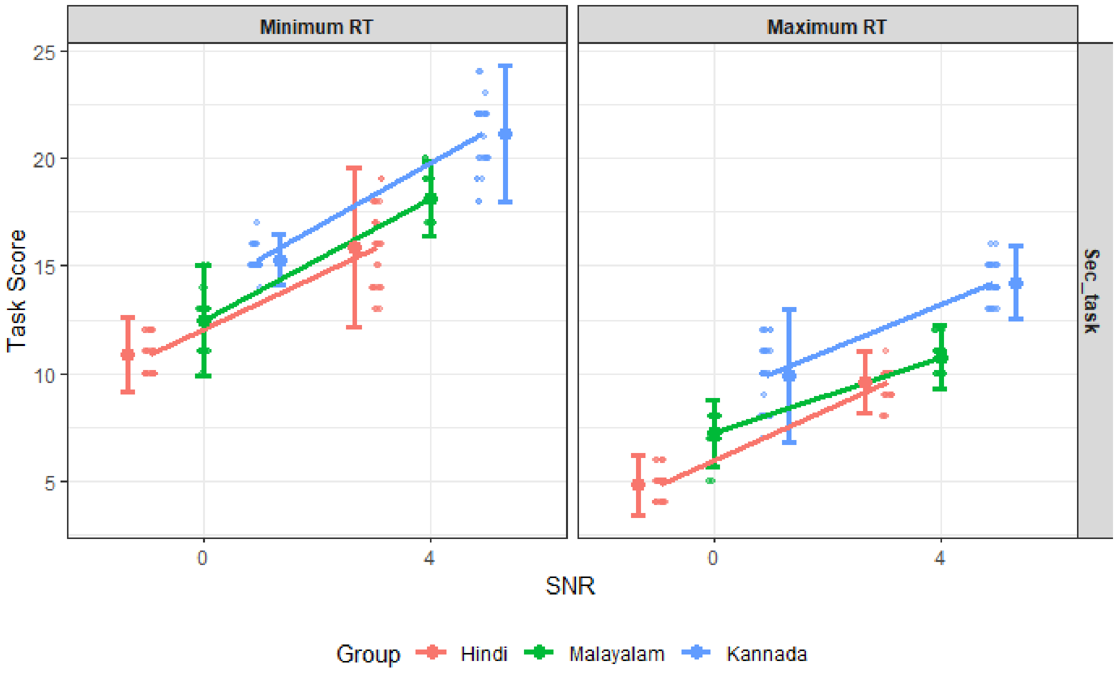
Abstract

Background. Speech perception in noise requires substantial cognitive load, yet listening effort remains underexplored, particularly in multilingual contexts like India where linguistic factors interact with acoustic challenges. This study examined listening effort in native and non-native Kannada speakers across varying signal-to-noise ratios (SNR) and reverberation conditions.

Methods. A mixed factorial design was used with 90 adults (18–40 years) divided into three groups: Hindi L1 (Group 1), Malayalam L1 (Group 2), and native Kannada speakers (Group 3). Listening effort was assessed using a dual-task paradigm. Kannada sentences mixed with multi-talker babble were presented at two SNRs (0 and +4 dB) across three reverberation times (RT1: 0.74s, RT2: 1.66s, RT3: 4.37s). 5 Kannada sentences were played in a block, and participants were asked to recall last but one word of each sentence in the block. Responses were recorded and analyzed using LMER in R Studio.

Results. Significant main effects of RT, SNR, and Group were observed ($p < .001$). Increased reverberation reduced performance ($\beta = -6.03$), while higher SNR improved scores ($\beta = +4.97$). Native Kannada speakers outperformed non-native groups, with highest listening effort in Hindi speakers followed by Malayalam speakers. A significant RT × SNR × Group interaction ($p < .001$) showed that SNR benefits decreased under high reverberation, especially for non-native listeners. Malayalam speakers performed better than Hindi speakers, likely due to greater linguistic similarity to Kannada.

Conclusions. Listening effort is influenced by both acoustic conditions and linguistic background. Native proficiency reduces cognitive load, whereas non-native listeners experience greater difficulty under degraded conditions, highlighting implications for audiology and multilingual communication.


Minimum RT
Maximum RT
Sec_task
Task Score
25
20
15
10
5
0
4
SNR
Group
Hindi
Malayalam
Kannada

HD-fNIRS Assessment of Listening Effort in CycleGAN-Enhanced Virtual Acoustics

Ali Syed (Hörzentrum Oldenburg gGmbH)*; Kirsten Wagener (Hörzentrum Oldenburg gGmbH); Volker Hohmann (Carl von Ossietzky Universität Oldenburg); Michael Richter (Liverpool John Moores University)

syed@hz-ol.de

Podium

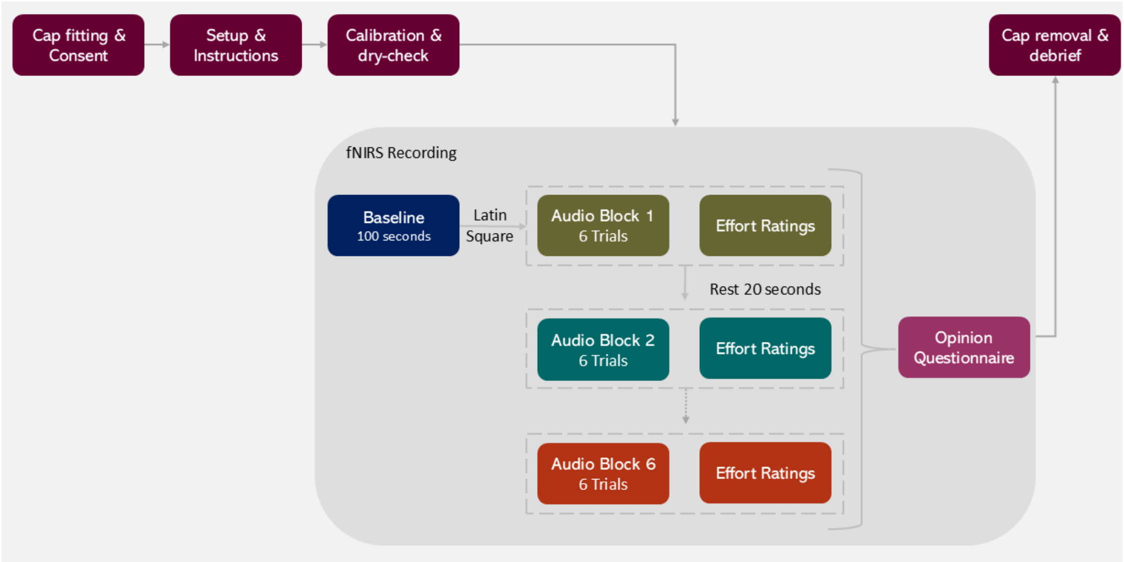
Abstract

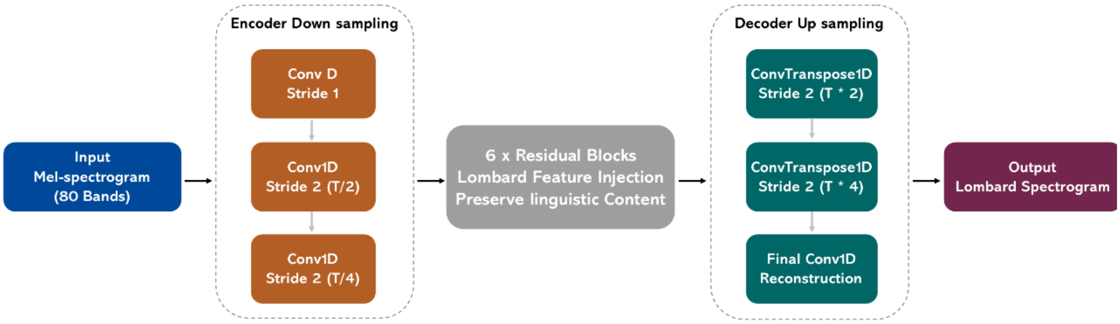
Background. Standard clinical assessments lack ecological validity to measure cognitive load in complex soundscapes. To address this, we developed a systematic framework integrating Virtual Acoustics (VA), AI speech synthesis, and neuroimaging to objectively quantify listening effort.

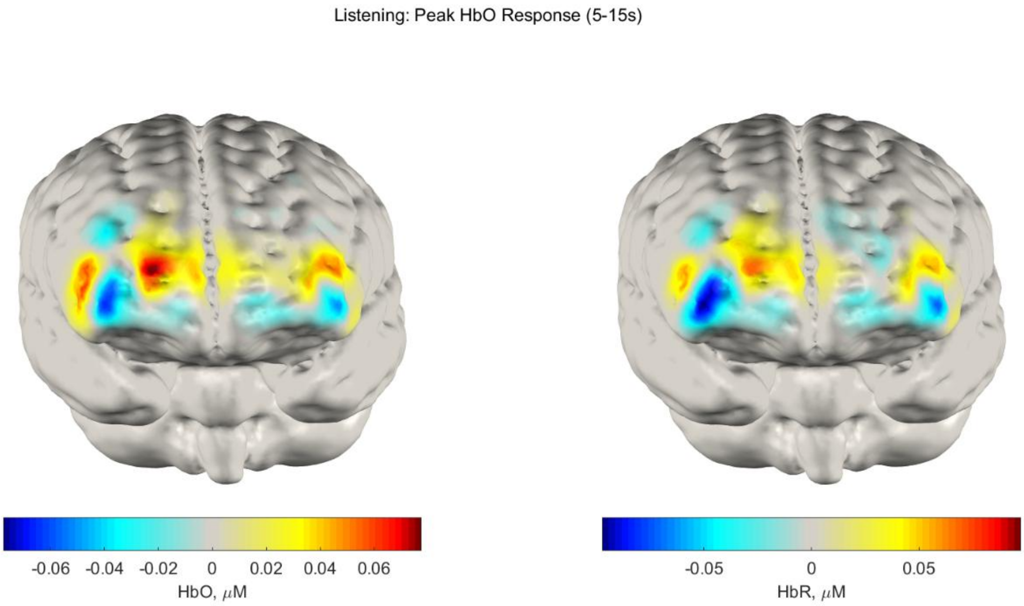
Methods. Three occupational environments were simulated in TASCAR, enhancing published models with custom material reflectivities and crowd sources. Delivered via HRTF binaural rendering, dynamic transitions utilize 300ms soft-ramp gain interpolation to prevent startle artifacts. Target stimuli comprised 36 sentences synthesized via ElevenLabs. To simulate vocal exertion while preserving RMS levels, a 1-D Convolutional CycleGAN (6 residual blocks) processed 80-band Mel-spectrograms to inject Lombard features (SNR -8 to -12 dB). Reconstructed via Griffin-Lim to prevent vocoder artifacts, stimuli maintained high intelligibility (STOI >0.85). Neutral speech formed the low-demand condition (SNR +2 to +6 dB). Using a Latin Square design, testing is ongoing (target N=48). Cortical hemodynamics are captured via HD-fNIRS. An adaptive Recursive Least Squares (RLS) filter isolates neural signals, followed by Diffuse Optical Tomography (DOT) for 3D reconstructions. Subjective and behavioral metrics are concurrently recorded.

Results. Technical validation confirmed <5ms audio-to-physiological latency; behavioral scores validated high VA ecological validity. Preliminary analysis indicates CycleGAN Lombard conditions elicit higher hemodynamic activation than baseline, with mean peak HbO values from 0.108 to 0.130 µM. Early DOT reconstructions demonstrate localized cortical responses correlate strongly with subjective effort and NASA-TLX scores.

Conclusions. Integrating VA, CycleGAN speech, and DOT yields a robust computational pipeline for evaluating listening effort, paving the way for advanced assessments of hearing interventions in ecologically valid environments.


Cap fitting & Consent
Setup & Instructions
Calibration & dry-check
Cap removal & debrief
fNIRS Recording
Baseline
100 seconds
Latin Square
Audio Block 1
6 Trials
Effort Ratings
Rest 20 seconds
Audio Block 2
6 Trials
Effort Ratings
Opinion Questionnaire
Audio Block 6
6 Trials
Effort Ratings


Input
Mel-spectrogram
(80 Bands)
Encoder Down sampling
Conv D
Stride 1
Conv1D
Stride 2 (T/2)
Conv1D
Stride 2 (T/4)
6 x Residual Blocks
Lombard Feature Injection
Preserve linguistic Content
Decoder Up sampling
ConvTranspose1D
Stride 2 (T * 2)
ConvTranspose1D
Stride 2 (T * 4)
Final Conv1D
Reconstruction
Output
Lombard Spectrogram


Listening: Peak HbO Response (5-15s)
-0.06 -0.04 -0.02 0 0.02 0.04 0.06
HbO, μM
-0.05 0 0.05
HbR, μM

Infants behave like adults during speech-in-noise processing—but what happens in the brain?

Irene Arrieta (Universite Paris Cite, INCC)*

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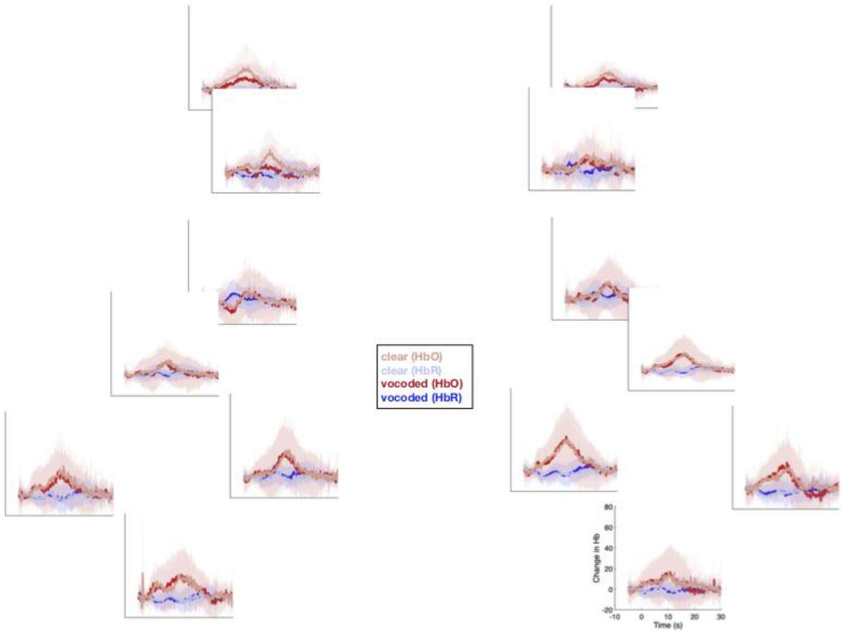
Podium

Abstract

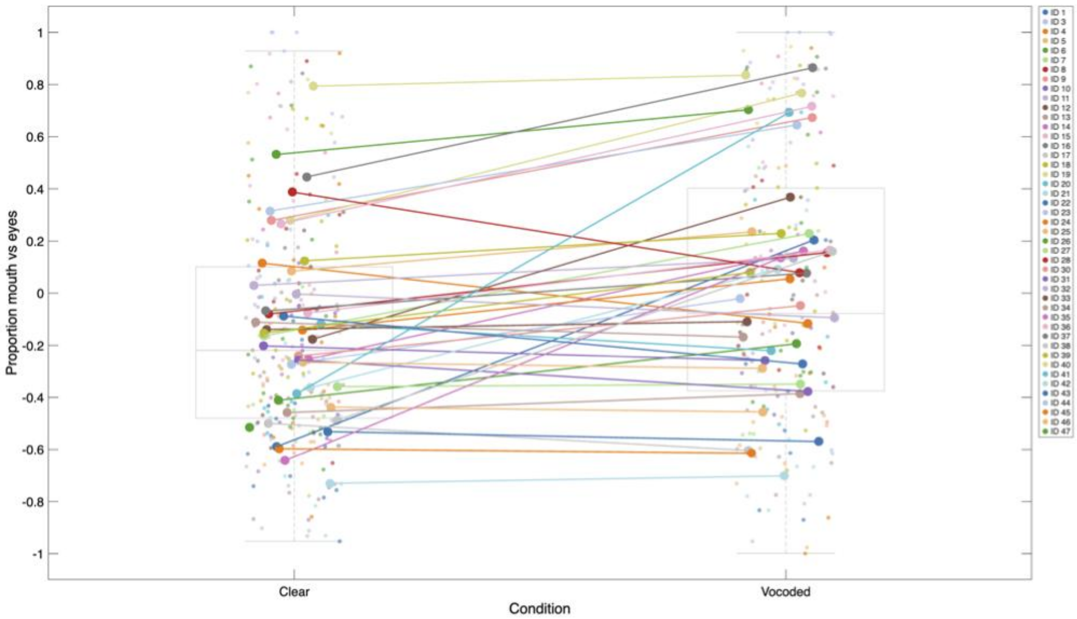
Background. Language acquisition typically occurs in rich audio-visual, face-to-face interactions, often under suboptimal listening conditions. Yet, much research on early language development has relied on auditory-only paradigms in ideal listening environments. In this study, we investigated how infants process audio-visual speech when intelligibility is reduced. We hypothesised that reduced speech intelligibility would: (1) Increase infants' attention to the mouth of the speaker; (2) Increase hemodynamic activity in brain areas related to executive functions and processing effort, such as the prefrontal cortex; and (3) Decrease cortical tracking of speech.

Method: We tested 47 infants aged 7–9 months, a developmental period characterized by a shift in visual attention from the eyes to the mouth during speech perception. Infants viewed 28 videos of a woman producing continuous infant-directed speech (23–30 s each) with either intact audio or degraded audio (8-band vocoding). During a 12-min session, gaze behavior and neural activity were recorded simultaneously. Eye tracking was used to assess visual attention to the mouth over the eyes of the speaker, while brain responses were measured using functional near-infrared spectroscopy (fNIRS) to index hemodynamic changes in fronto-temporal regions and electroencephalography (EEG) to quantify neural-speech synchrony using cortical tracking analysis.

Results and Conclusion: Behavioral results showed that infants significantly increased looking time toward the speaker's mouth when speech was degraded, suggesting an early compensatory strategy similar to that observed in adults. Analysis of the fNIRS and EEG data is still ongoing, with completion planned before the conference and therefore the second and third hypotheses remain to be statistically tested. This research aims to establish objective measures of speech intelligibility in young listeners, with potential applications for infants with hearing loss.



Group-level hemodynamic responses by condition across optode channels covering frontotemporal brain areas. Red and light red lines represent oxygenated hemoglobin (HbO) concentration changes across trials, whereas blue and light blue lines represent deoxygenated hemoglobin (HbR) concentration changes. Intense colors indicate the *vocoded* condition, while lighter colors indicate the *clear* (unmodified audio) condition. Shaded areas surrounding the lines represent the group standard deviation.



Proportion of total looking time directed to the mouth relative to the eyes across conditions. Positive values on the y-axis indicate a preference for looking at the mouth, whereas negative values indicate a preference for looking at the eyes. Large colored dots represent each participant's mean proportion across trials, while small dots represent individual trial values. The *clear* condition corresponds to videos presented with unmodified audio, whereas the *vocoded* condition corresponds to the same videos presented with 8-channel vocoded audio. tone-vocoder.

Corticothalamic Mechanisms of Human Auditory Selective Attention

Ana-Maria Gore (Maastricht University)*; Ryszard Auksztulewicz (Maastricht University); Michelle Moerel (Maastricht University)

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Poster

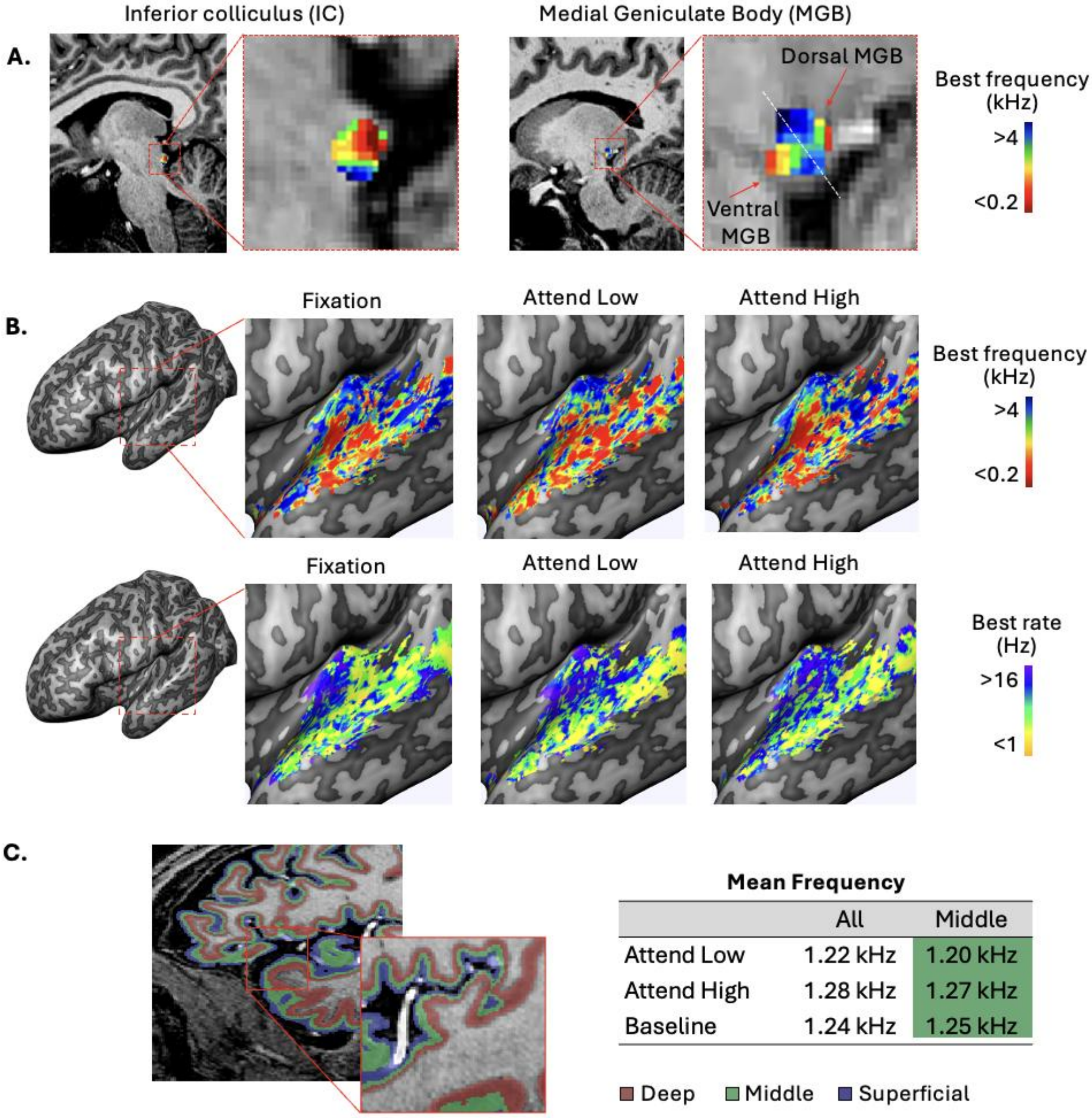
Abstract

Background. Selective auditory attention allows humans to focus on a relevant sound stream while ignoring competing noise. However, the neural mechanisms underlying this capacity remain poorly understood. Animal studies have implicated the corticothalamic circuitry in attention, but whether these findings extend to humans remains unexplored.

Methods. To address this, ultra-high field (7T) fMRI was collected while participants performed an auditory selective attention task. Participants attended to tone frequency (low vs. high) or temporal modulation rate (slow vs. fast) and performed a gap detection task. Using this paradigm, we map cortical depth-dependent responses in auditory cortex and in subcortical nuclei, including the medial geniculate body (MGB) and inferior colliculus. With population receptive field modelling (pRF), we characterise tonotopic and temporal tuning profiles (preferred feature and tuning width) and test whether attention shifts tuning preferences or modulates response gain across cortical depths and subcortical structures.

Results. Pilot 7T fMRI data (N = 1) show tonotopic maps and modulation rate preference separately in deep, middle, and superficial cortical depths. Subcortically, frequency-specific responses were observed in both MGB and inferior colliculus with MGB responses allowing segregation into ventral and dorsal divisions. While we observe shifts in frequency preference with attention in this participant, more data is needed to evaluate the robustness of this finding.

Conclusions. These findings demonstrate the feasibility of collecting fMRI data in auditory subcortical structures and across cortical depth within the same acquisition, allowing the exploration of the role of the corticothalamic circuitry in human auditory selective attention. Data collection and analysis are ongoing.



Mean Frequency

	All	Middle
Attend Low	1.22 kHz	1.20 kHz
Attend High	1.28 kHz	1.27 kHz
Baseline	1.24 kHz	1.25 kHz

Figure 1: Attention-induced changes in frequency and temporal modulation rate preference.
A. Subcortical tonotopy in the inferior colliculus (IC) and medial geniculate body (MGB), with a frequency reversal delineating ventral from dorsal MGB.
B. Cortical maps of preferred frequency (tonotopy; top) and temporal modulation rate (bottom) across conditions, showing largely similar topography with subtle differences across conditions.
C. Cortical depth-dependent sampling using equi-volume surfaces allows mapping condition-specific differences in mean preference across cortical depth. Pilot data (N=1) show a shift of toward the attended frequency emerging already in the middle layers of primary auditory cortex (PAC).

Effects of Environmental Auditory Complexity on First- and Higher-Order Mismatch Negativity: A Test of Hierarchical Predictive Processing

Sahana Murali (Empathic Computing Lab)*

sahanamurali.official@gmail.com

Poster

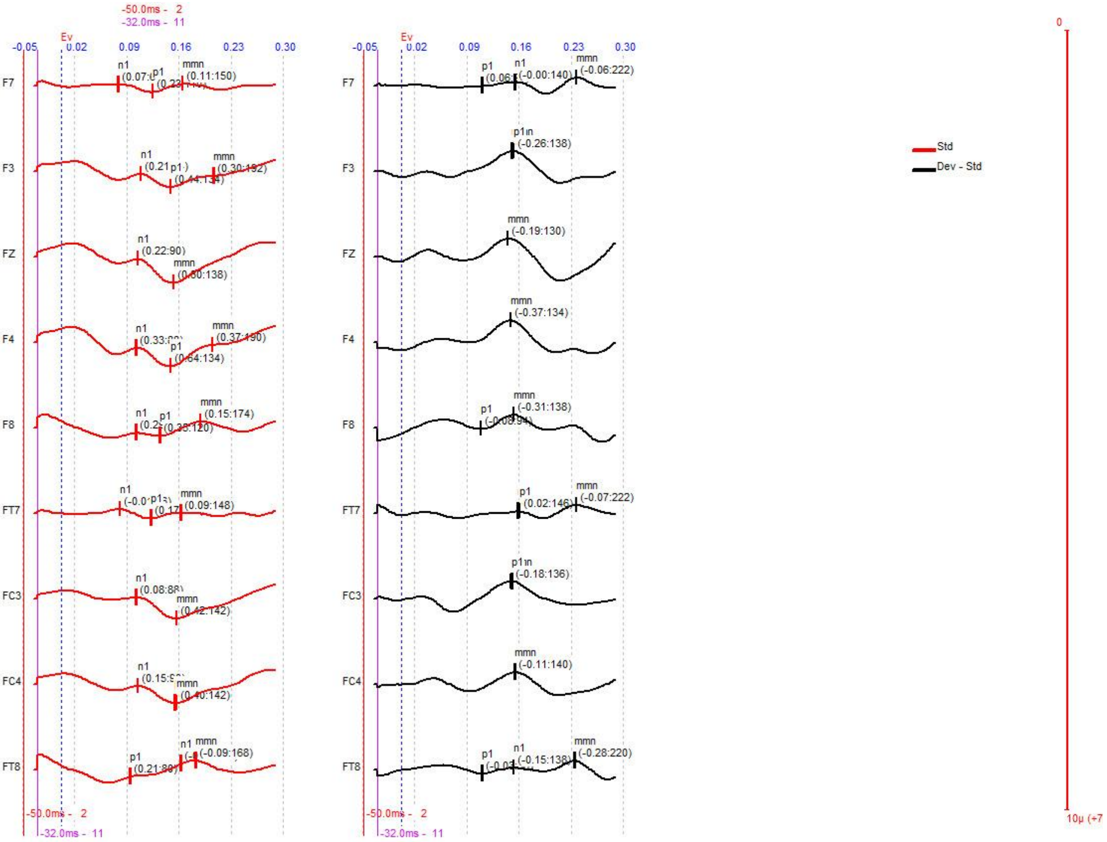
Abstract

Background. Mismatch negativity (MMN) is a well-established neural index of prediction error in auditory processing. Predictive processing accounts propose that such predictions are hierarchically organised; however, it remains unclear how different levels of prediction are affected by increasing environmental complexity. In particular, it is unknown whether auditory entropy differentially disrupts first-order (global probability-based) and higher-order (transition-based) predictive mechanisms.

Methods. EEG is recorded during passive listening in a within-subject design manipulating MMN type (first-order vs higher-order) and environmental complexity (low vs high). Complexity is operationalised by increasing the number of filler tones while maintaining constant deviant probability, thereby isolating entropy effects independent of violation rate. First-order MMN is elicited using an oddball paradigm, whereas higher-order MMN is elicited via violations of conditional transition rules. An equiprobable control condition is included to dissociate prediction error from sensory adaptation. N1 (50–150 ms) and MMN (100–250 ms) amplitudes will be analysed at fronto-central electrodes.

Results. Data collection is ongoing. It is hypothesised that increased environmental complexity will attenuate MMN amplitude, reflecting reduced predictive precision. A concurrent reduction in N1 amplitude would indicate degraded sensory encoding, whereas stable N1 alongside reduced MMN would suggest selective down-weighting of prediction error signals. Higher-order MMN is expected to show greater sensitivity to complexity than first-order MMN.

Conclusions. This study dissociates hierarchical levels of auditory prediction while controlling for adaptation and violation probability. It tests whether increasing environmental complexity reduces the magnitude of prediction error responses and whether higher-order predictive mechanisms are more susceptible to such effects.



F7
F3
FZ
F4
F8
FT7
FC3
FC4
FT8
Std
Dev - Std
-50.0ms - 2
-32.0ms - 11
Ev
0
10µ (+7

Extending Dynamic Causal Modelling with Interlaminar Connectivity for Laminar fMRI

Alicia Gonsalves (Maastricht University)*; Ryszard Auksztulewicz (Maastricht University); Michelle Moerel (Maastricht University)

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Poster

Abstract

Background. Laminar fMRI offers a noninvasive view of cortical microcircuits, but interpreting these signals is challenging, as BOLD reflects hemodynamic rather than neuronal activity directly. This requires computational models that can disambiguate this relationship. Within Dynamic Causal Modelling (DCM), this is achieved in P-DCM by pairing a neuronal model with a hemodynamic model correcting for ascending vein effects in laminar BOLD. Yet, current work models the neuronal layers without interlaminar connectivity essential to cortical processing. We extend this framework by including interlaminar connections to capture cortical information flow.

Methods. We validate a single-region P-DCM comprising a neuronal model (N=3 layers), neurovascular coupling, and a hemodynamic model (K=7 depths) (Fig. 1). Forward simulations establish face validity, and parameter recovery assesses identifiability of interlaminar connectivity (A), modulation (B), driving input (C), and neuronal parameters (σ, μ, λ).

Results. Simulations show plausible layer-specific dynamics and distinct BOLD depth profiles (Fig. 2). Following driving input, the middle (M) layer responded first, propagating activity to superficial (S) and deep (D) layers via A. Modulation increased responses globally. Parameter recovery from noise-free simulations showed partial identifiability (Fig. 3). C, A (M→S), A (S→D), and μ recovered well. B mirrored A but degraded: M→S recovered well; S→D and D→M showed reduced accuracy. σ and λ showed range-dependent recovery. D→M showed flat recovery for A and B.

Conclusions. These results show that the model captures core dynamics and clarifies identifiability boundaries. Poor recovery of D→M and λ likely reflects BOLD's limited temporal resolution, inadequately constraining fast neuronal dynamics. Next steps involve empirical 7T fMRI validation and Bayesian fusion with iEEG. The model will be the foundation for a multi-region framework to study auditory microcircuit dynamics.

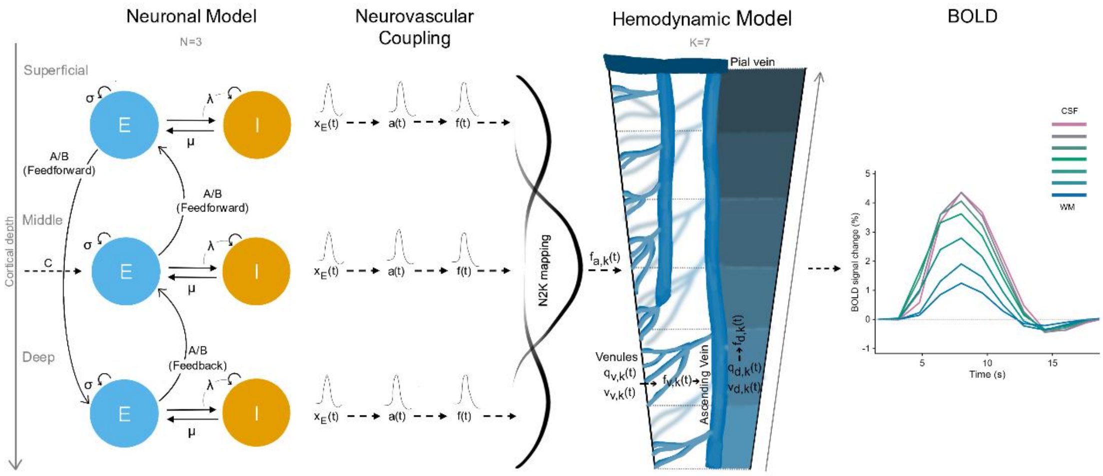
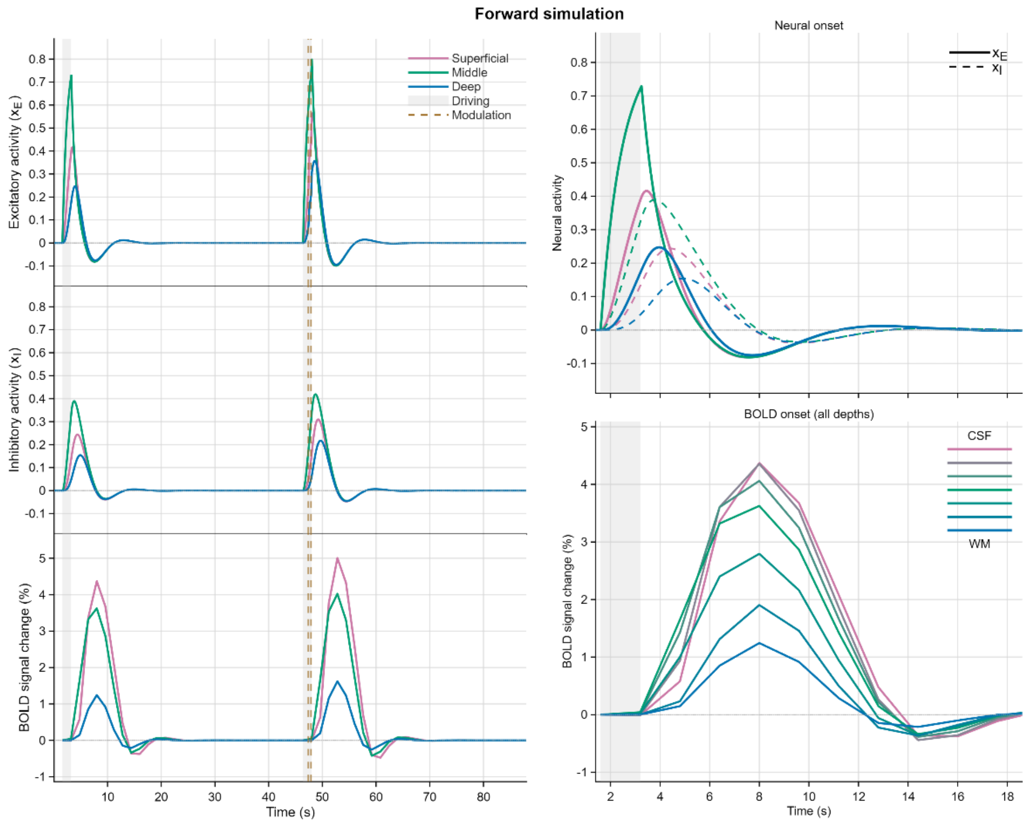


Figure 1. Schematic of the integrated laminar P-DCM framework. The neuronal model (left) is an adaptive dynamical system with N=3 cortical layers, each containing coupled excitatory (E) and inhibitory (I) populations. Layer dynamics are governed by excitatory self-decay (σ), inhibitory-to-excitatory connection (μ), and inhibitory gain (λ). λ controls both the amplitude and temporal smoothness of inhibitory responses. Interlaminar feedforward and feedback connections (A) allow directional information flow across layers; modulatory connectivity (B) scales these connections. Driving input (C) targets the middle layer. Neurovascular coupling (NVC) translates layer-specific excitatory activity $x_E(t)$ into a vasoactive signal a(t) and blood flow response f(t). A N2K mapping distributes neuronal blood flow f(t) across K=7 vascular depths as $f_{a,k}(t)$, with each depth receiving weighted contributions from neighbouring layers. The hemodynamic model tracks blood volume v(t) and deoxyhaemoglobin q(t) across microvasculature (v) and ascending vein compartments (d) at each depth k. Blood draining upward through the ascending vein accumulates signal from deeper layers, producing the surface bias in the predicted laminar BOLD response (right). P-DCM = Physiologically informed DCM.



N = 3 layers, K = 7 depths | $A_{mid \to sup}$ = 1.00, $A_{sup \to deep}$ = 1.00, $A_{deep \to mid}$ = 1.00, C_{mid} = 1.00
σ = -1.00, μ = -0.80, λ = 1.80, α_d = 0.00, s_d = 0.00
$B_{mid \to sup}$ = 1.00, $B_{sup \to deep}$ = 1.00, $B_{deep \to mid}$ = 1.00

Figure 2. Face validity of forward simulation. Left column shows excitatory (x_E) and inhibitory (x_I) neuronal activity and BOLD signal change across superficial, middle, and deep layers for the full simulation epoch. Grey shading indicates driving input periods and dashed lines indicate the onsets of modulatory input. Driving input to the middle layer produces the earliest and strongest response, with activity sequentially propagating to superficial and then deep layers via interlaminar connections. Inhibitory activity mirrors excitatory responses with a slight delay, reflecting intralaminar E→I coupling. Modulatory input increases the neural response in the second event. Right column shows zoomed views of the neural onset (top) and the BOLD response across all seven vascular depths (bottom) for the first stimulus event. BOLD amplitude is largest near the cerebrospinal fluid (CSF) boundary and decreases toward white matter (WM), consistent with ascending vein drainage. Simulation parameters are shown above.

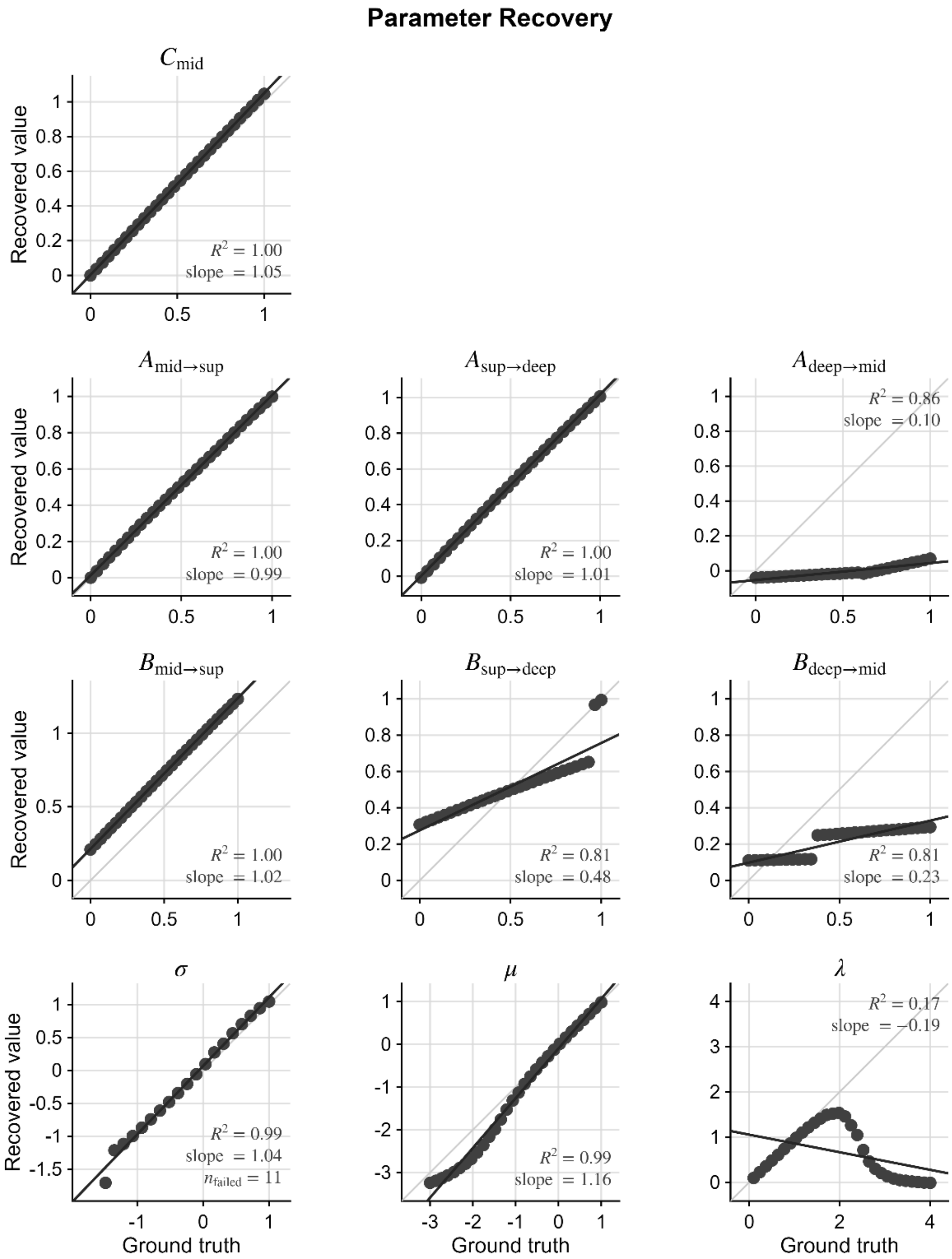


Figure 3. Parameter recovery from noise-free simulations. Scatter plots of ground truth versus recovered parameter values for each estimated parameter, based on 30 simulations equally spaced within a plausible range. The diagonal line indicates perfect recovery. R^2 and slope are shown for each parameter. n_{failed} indicates the number of simulations excluded due to numerical instability during integration. Top row: driving input (C). Second row: interlaminar connectivity (A). Third row: modulatory connectivity (B). Bottom row: neuronal dynamics (σ, μ, λ).

Autonomic Responses to Affective Sounds in Misophonia and Hyperacusis

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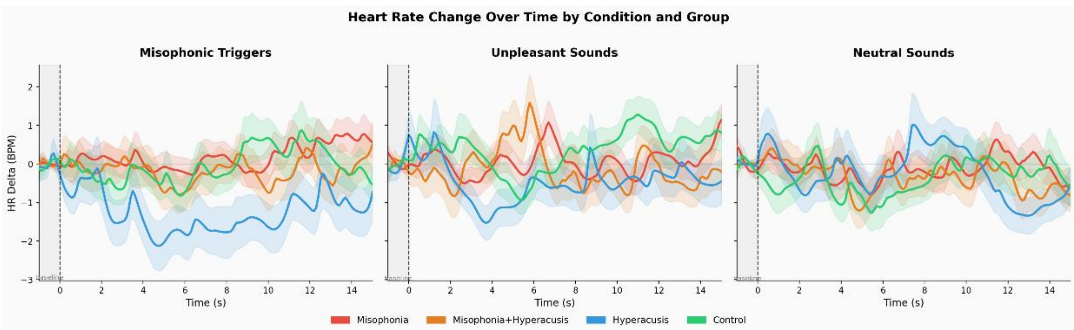
Poster

Abstract

Sounds evoke affective responses measurable via physiological changes such as heart rate and respiration. These measures may help characterize sound tolerance disorders such as misophonia and hyperacusis, where every day sounds elicit disproportionate reactions. Misophonia is characterized by intense emotional responses to specific trigger sounds, whereas hyperacusis involves heightened sensitivity to sound loudness. Although phenomenologically distinct, these conditions frequently co-occur. Yet, their autonomic correlates remain poorly understood and have not been directly compared. This study examined heart rate (HR) and respiration (RSP) during fMRI across two affective sound-rating paradigms: a trigger task (misophonic, unpleasant, neutral) and an IADS task (pleasant, unpleasant, neutral). Eighty-seven participants were divided into four groups: Misophonia (M, n=28), Misophonia & Hyperacusis (MH, n=24), Hyperacusis (H, n=14), and Controls (C, n=21). Data was preprocessed using NeuroKit2 and epoched from 1 to 16s (trigger task) and -1 to 6s (IADS task). Mixed ANOVAs revealed a significant group effect for RSP ($F(3,81)=2.92$, $p=.039$). The Misophonia group showed decreased respiratory rate compared to Controls; the Hyperacusis group showed increased respiration. Time-series analyses showed the Hyperacusis group exhibited the largest HR decreases during misophonic sounds, peaking ~4s post-stimulus (mean=-1.46 BPM), though this did not reach significance ($F(3,83)=1.85$, $p=.145$), likely due to limited sample size (n=14). For the IADS task, within-group analyses revealed a significant condition effect driven primarily by the Hyperacusis group ($F(2,26)=4.07$, $p=.029$, $\eta^2=.112$), with a pronounced HR decrease during pleasant sounds. No significant condition effects emerged for the other groups. These findings suggest distinct autonomic profiles across sound sensitivity disorders, with the Hyperacusis group showing the strongest physiological reactivity across both paradigms.



Session 3.A. Hidden Hearing Loss and Subclinical Auditory Disorders

From synaptopathy to auditory fatigue: mechanisms, models, and diagnostic markers.

Chaired by Dr. Joaquin T. Valderrama.

Efficient Coding Explains Altered Neural Representations in Hidden Hearing Loss

Juan M. Fuentes (Universitat Pompeu Fabra)*

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Featured talk

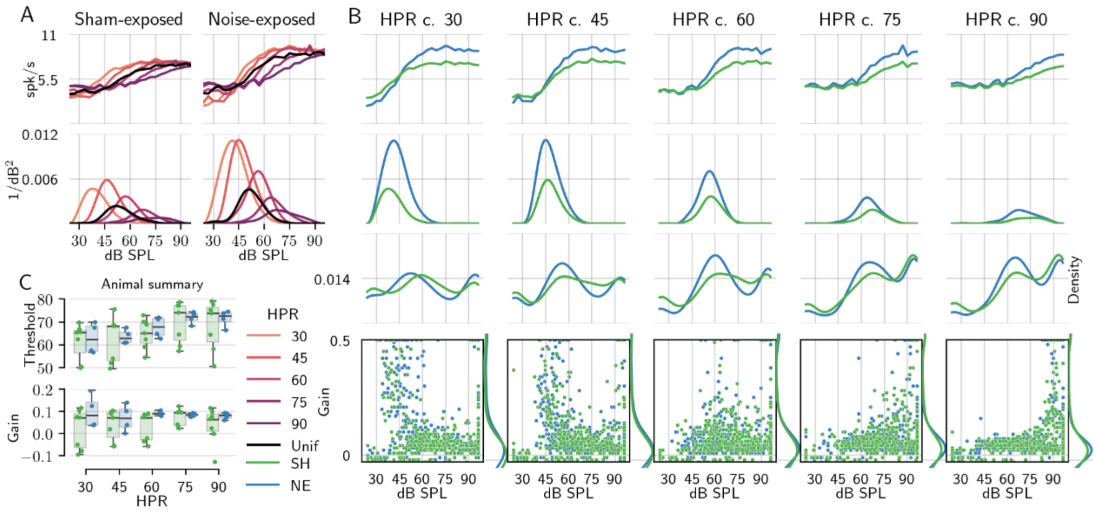
Abstract

Background. Sensory systems must represent a vast range of stimulus energy under metabolic constraints. Efficient-coding theory predicts that adaptation re-allocates limited neural activity toward informative stimulus values, but it remains unclear how hidden hearing loss shifts this operating point in central auditory circuits. Hearing is a stringent test because sound level varies enormously across environments, yet clinical assessment still relies on thresholds that can miss listening deficits in noise.

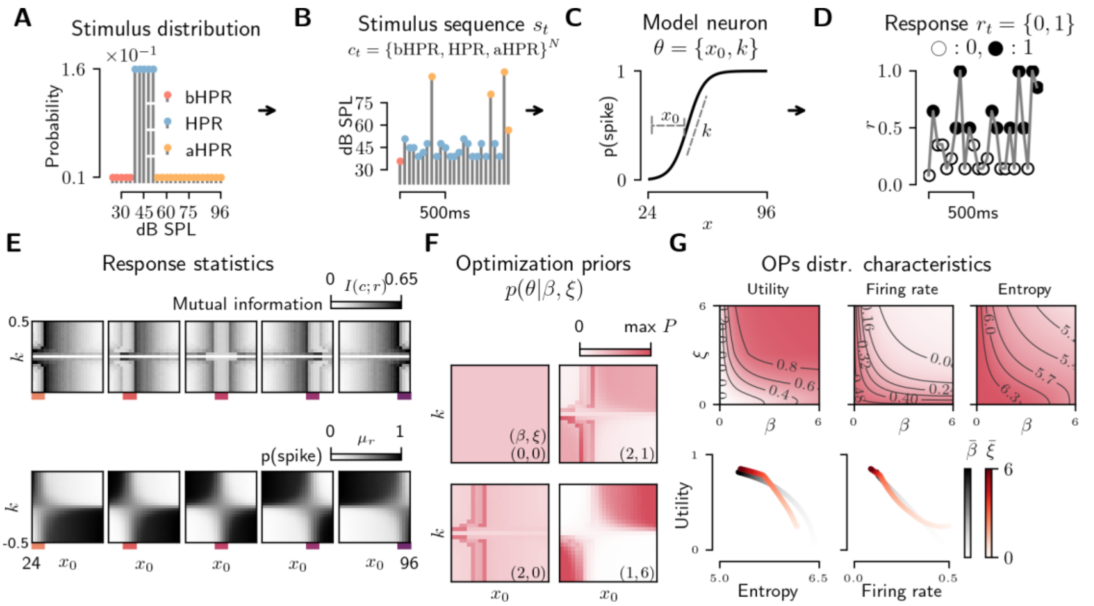
Methods. We analyzed extracellular recordings from single neurons in the gerbil auditory midbrain across 14 animals in four experimental groups exposed to sound-level distributions drawn either uniformly from 24-96 dB SPL or from contexts in which 80% of levels fell within a 12-dB high-probability range. For each context, we summarized each neuron's rate-intensity function by effective threshold and gain, then interpreted threshold-gain distributions with an information-cost model trading stimulus information against mean spiking.

Results. Noise exposure consistent with hidden hearing loss altered gain modulation across acoustic contexts. Noise-exposed animals showed compressed gain adjustments relative to controls; within the information-cost framework, the clearest effect was a quiet-context utility advantage concentrated in low-threshold neurons, whereas moderate-to-loud contexts showed weaker or absent group differences. Temporary conductive attenuation shifted effective thresholds to higher sound levels, with incomplete recovery after plug removal.

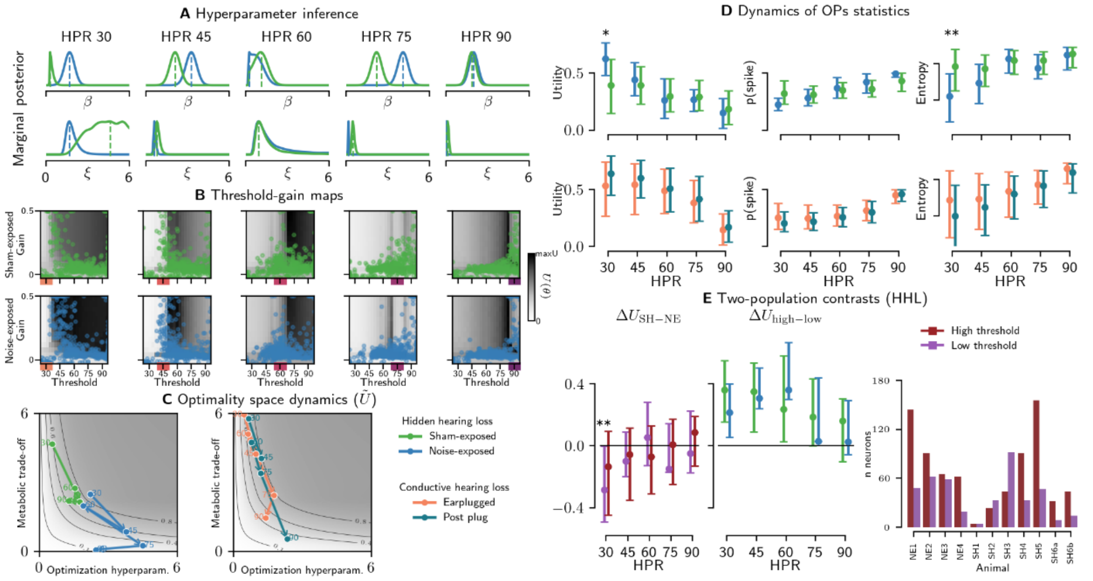
Conclusions. These results support an efficient-coding interpretation of altered central auditory representations in hidden hearing loss and provide a context-based framework for understanding hearing difficulty beyond threshold-only tests and Fisher information alone.



A
Sham-exposed
Noise-exposed
B
HPR c. 30
HPR c. 45
HPR c. 60
HPR c. 75
HPR c. 90
spk/s
1/dB²
dB SPL
C
Animal summary
Threshold
Gain
HPR
30
45
60
75
90
Unif
SH
NE
Density



A
Stimulus distribution
Probability
bHPR
HPR
aHPR
dB SPL
B
Stimulus sequence s_t
$c_t = \{\text{bHPR}, \text{HPR}, \text{aHPR}\}^N$
dB SPL
500ms
C
Model neuron
$\theta = \{x_0, k\}$
p(spike)
D
Response $r_t = \{0, 1\}$
500ms
E
Response statistics
Mutual information
p(spike)
F
Optimization priors
$p(\theta|\beta, \xi)$
max P
G
OPs distr. characteristics
Utility
Firing rate
Entropy
Utility
Entropy
Firing rate



A Hyperparameter inference
HPR 30
HPR 45
HPR 60
HPR 75
HPR 90
Marginal posterior
B Threshold-gain maps
Sham-exposed Gain
Noise-exposed Gain
Threshold
C Optimality space dynamics ($\bar{U}$)
Metabolic trade-off
Optimization hyperparam.
Hidden hearing loss
Sham-exposed
Noise-exposed
Conductive hearing loss
Earplugged
Post plug
D Dynamics of OPs statistics
Utility
p(spike)
Entropy
HPR
E Two-population contrasts (HHL)
$\Delta U_{\text{SH-NE}}$
$\Delta U_{\text{high-low}}$
High threshold
Low threshold
n neurons
Animal

Objective SNR Benefits from Hearing Aids Fitted with the NAL-NL3 MHL Module: Effects of Performance Level and Dome Type

Gilles Courtois (Sonova AG)*

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Podium

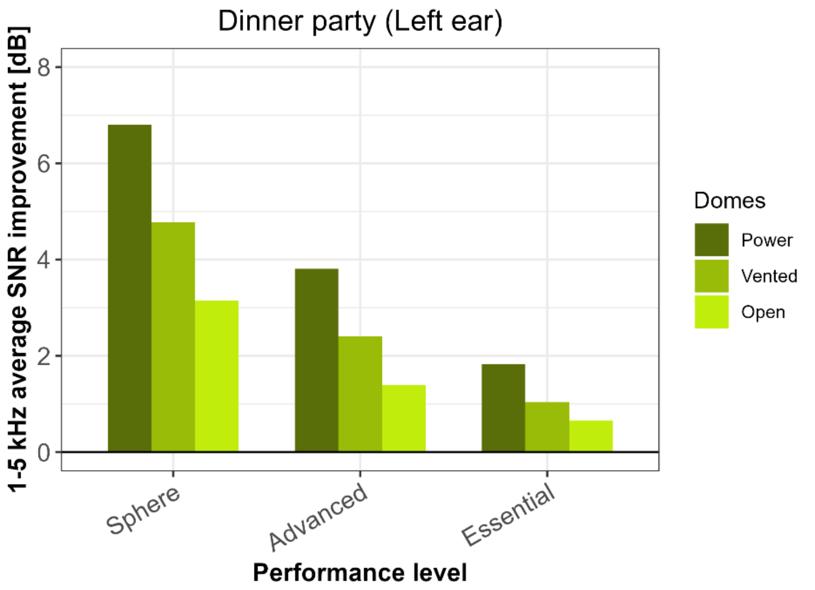
Abstract

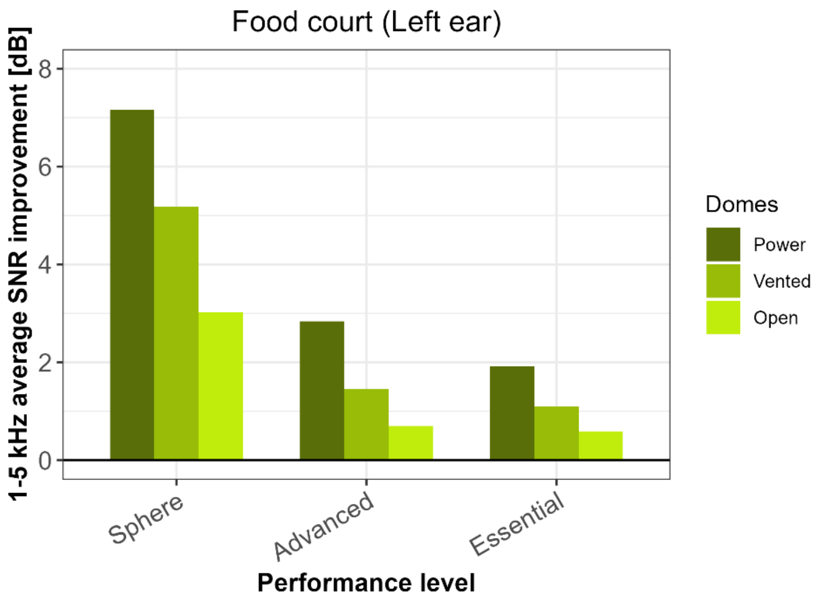
Background. Over the past decade, there has been growing evidence that open-fit, low-gain hearing aids can benefit individuals with subclinical hearing loss (SCHL). In 2025, the National Acoustic Laboratories (NAL, Sydney, Australia) released their new fitting formula, the NAL-NL3, which includes a module specifically designed for SCHL. The Minimal Hearing Loss (MHL) module prescribes insertion gain that is partly independent of the audiogram and assumes occluding ear coupling to maximize access to directional microphones and noise reduction features embedded in current hearing devices. This represents a significant shift in the audiology field, as hearing aids have traditionally been intended only for those with clinical losses, and open fittings are commonly used in patients with the slightest degree of hearing loss.

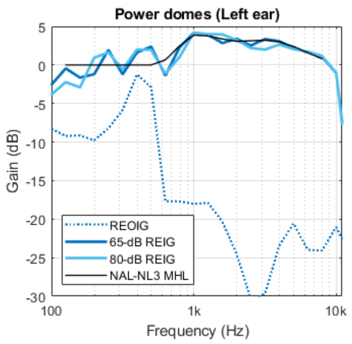
Methods. In this investigation, objective SNR benefits obtained with a pair of hearing aids fitted on a KEMAR using the NAL-NL3 MHL module were derived using the Hagerman & Olofsson method. Two higher-order Ambisonic scenes from the ARTE database were reproduced over a 12-loudspeaker array, and target speech samples recorded at realistic vocal effort were used. The effects of three different noise reduction features, corresponding to three different price points, and three dome types (open, vented, and power) were analyzed.

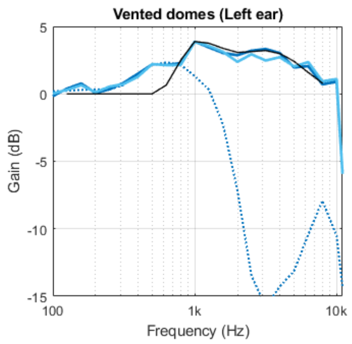
Results. The results show that the most advanced performance level (AI-based noise reduction) yields up to 4 dB greater SNR improvement than the entry-level feature (a single-channel, deterministic noise reduction). The results also confirm that a significant portion of the SNR benefit is lost when moving from power to open domes, supporting NAL's recommendation to fit the MHL module with the most occluding domes that can be tolerated by the end user.

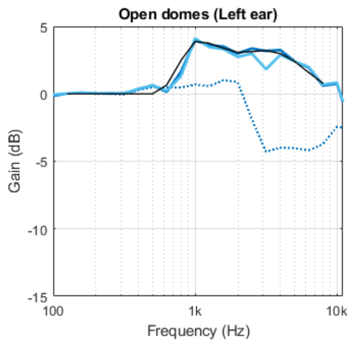
Conclusions. Overall, the findings of this investigation suggest that the new NAL-NL3 MHL module is a promising fitting tool to support speech understanding in loud and noisy environments for individuals with SCHL.


Dinner party (Left ear)
1-5 kHz average SNR improvement [dB]
8
6
4
2
0
Sphere
Advanced
Essential
Performance level
Domes
Power
Vented
Open


Food court (Left ear)
1-5 kHz average SNR improvement [dB]
8
6
4
2
0
Sphere
Advanced
Essential
Performance level
Domes
Power
Vented
Open


Power domes (Left ear)
Gain (dB)
Frequency (Hz)
100
1k
10k
REOIG
65-dB REIG
80-dB REIG
NAL-NL3 MHL


Vented domes (Left ear)
Gain (dB)
Frequency (Hz)
100
1k
10k


Open domes (Left ear)
Gain (dB)
Frequency (Hz)
100
1k
10k

Phenotypic changes of auditory nerve fibers after excitotoxicity

Jérôme Bourien (Institute for Neurosciences of Montpellier)*

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Podium

Abstract

Background. Synaptic transmission between sensory inner hair cells (IHCs) and type I auditory nerve fibers (ANFs) is primarily glutamatergic. Excessive glutamate release—triggered by acoustic trauma or impaired clearance—leads to excitotoxicity, a neurotoxic state. Although the pathophysiology of this process in the mammalian cochlea is well-documented, the potential for long-term reversibility of the resulting neural damage remains a critical, unresolved question in auditory neuroscience.

Methods. To simulate a severe excitotoxic insult, we assessed the long-term structural and functional outcomes of kainate application (25 mM) in the gerbil cochlea (following the model described by Diuba et al., PNAS, 2025).

Results. Despite a permanent 40% reduction in IHC synapses across the entire tonotopic axis, the compound action potential (CAP) of the auditory nerve showed complete recovery. This functional restoration was underpinned by a significant phenotypic shift in the surviving ANFs. Specifically, fiber activation thresholds were enhanced globally. In the apical region, high-spontaneous rate (SR) fibers became the predominant population. In the basal region, surviving low-SR fibers underwent a functional transformation, displaying sound-driven activity that was virtually indistinguishable from control high-SR fibers.

Conclusion: Our findings indicate that a compensatory phenotypic adaptation in auditory neurons enables the full recovery of neural response thresholds and amplitudes, even in the presence of chronic synaptopathy. However, this peripheral hyper-responsiveness likely serves as a primary driver for central auditory hyperactivity, providing a mechanistic link between excitotoxic injury and the development of disorders such as tinnitus and hyperacusis.

Subclinical Effects of Recreational Noise Exposure: A Pre-registered Longitudinal Study

Hannah Guest (The University of Manchester); Rebecca Dewey (University of Nottingham); Karolina Kluk (The University of Manchester); Rebecca Millman (The University of Manchester); Kevin Munro (The University of Manchester); Carlyn Murray (The University of Manchester); Garreth Prendergast (The University of Manchester); Stephen Roberts (The University of Manchester); Chris Plack (The University of Manchester)*

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Podium

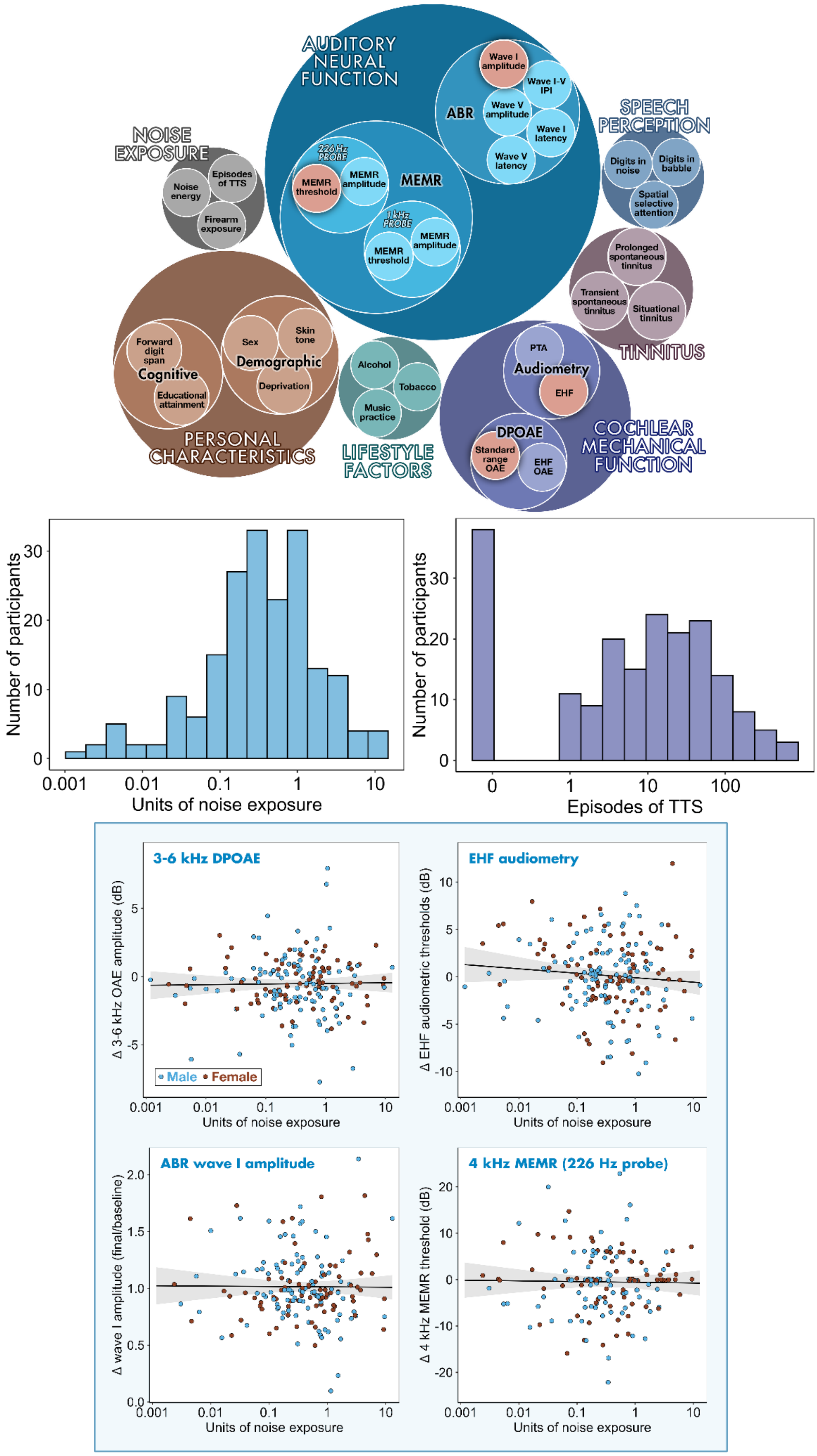
Abstract

Background. Findings on the subclinical effects of noise exposure have often been inconsistent and inconclusive, likely due to (a) small samples, (b) inaccurate retrospective estimates of noise exposure, (c) post-hoc data dredging, and (d) cross-sectional designs with large effects of between-subject variability. The present study used: a prospective approach with regular follow-ups (to improve estimates of noise exposure); repeated measures (to minimise unwanted between-subject variability); and pre-registration (to guard against data dredging).

Methods. 191 normally hearing teenagers (aged 16-17 at first timepoint) completed a 3-hour battery of physiological, behavioural, and self-report measures at two timepoints, 3 years apart (Fig. 1). Noise exposure was self-reported at 18-month intervals. The primary hypotheses addressed relations between noise exposure and longitudinal changes in: extended high-frequency audiometry; distortion-product otoacoustic emissions; auditory brainstem response wave I amplitude; and the middle-ear muscle reflex. Exploratory analyses included: changes in speech perception and tinnitus; and relation of outcomes to episodes of temporary threshold shift (TTS).

Results. Over the three years, participants reported a large range of noise exposures (equivalent to 0-690 100-dBA 2-hour concerts), and a large range of experiences of TTS (0-597 instances, median 11; Fig. 2). However, no associations were found between noise exposure or TTS and change in any outcome (primary or exploratory, physiological or behavioural). Figure 3 shows results relating to primary hypotheses.

Conclusions. Findings call for cautious reappraisal of the notion that recreational noise causes widespread subclinical cochlear damage in young people. The cohort provides a strong platform for longer follow-up to determine whether effects emerge over timescales not captured here.


AUDITORY NEURAL FUNCTION
ABR
Wave I amplitude
Wave I-V IPI
Wave V amplitude
Wave I latency
Wave V latency
MEMR
226 Hz PROBE
MEMR threshold
MEMR amplitude
1 kHz PROBE
MEMR threshold
MEMR amplitude
NOISE EXPOSURE
Noise energy
Episodes of TTS
Firearm exposure
SPEECH PERCEPTION
Digits in noise
Digits in babble
Spatial selective attention
TINNITUS
Prolonged spontaneous tinnitus
Transient spontaneous tinnitus
Situational tinnitus
PERSONAL CHARACTERISTICS
Cognitive
Forward digit span
Educational attainment
Demographic
Sex
Skin tone
Deprivation
LIFESTYLE FACTORS
Alcohol
Tobacco
Music practice
COCHLEAR MECHANICAL FUNCTION
Audiometry
PTA
EHF
DPOAE
Standard range OAE
EHF OAE
Number of participants
Units of noise exposure
Episodes of TTS
3-6 kHz DPOAE
Δ 3-6 kHz OAE amplitude (dB)
Male
Female
EHF audiometry
Δ EHF audiometric thresholds (dB)
ABR wave I amplitude
Δ wave I amplitude (final/baseline)
4 kHz MEMR (226 Hz probe)
Δ 4 kHz MEMR threshold (dB)

Extended high-frequency hearing predicts rapid spoken-word recognition

Tugba Lulaci (Lund University)*; Pelle Söderström (Lund University; National Acoustic Laboratories); Mikael Roll (Lund University)

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Podium

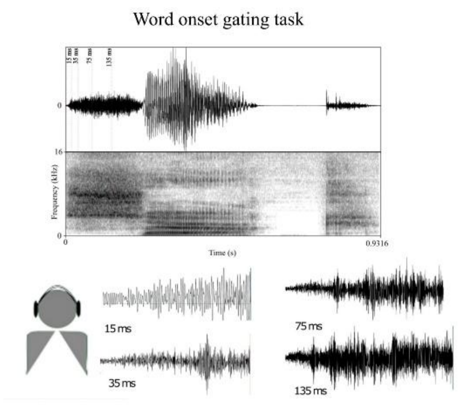
Abstract

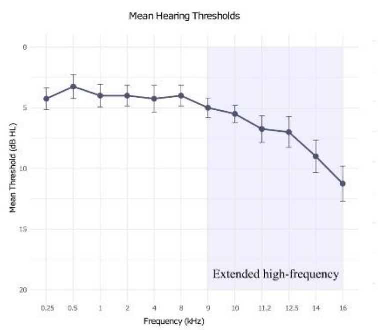
Background. Listeners with clinically normal audiograms can still vary in their spoken-word-recognition performance. Standard pure-tone hearing tests may overlook how meaningful hearing sensitivity variation contributes to these perceptual differences. Extended high-frequency (EHF; >8 kHz) hearing may support access to subtle acoustic cues that facilitate rapid spoken-word recognition.

Methods. This study examined the relationship between EHF thresholds and spoken-word recognition performance in an auditory gating task. Participants identified spoken words from increasingly longer word-onset fragments, allowing assessment of how efficiently early acoustic cues supported predictive spoken-word recognition.

Results. Listeners with better EHF thresholds were more accurate at early gates, successfully identifying words from as early as 15 ms of acoustic input when onsets contained more distinct cues and stronger high-frequency energy (e.g., /s/). This relationship was not observed for words with weaker or more diffuse high-frequency onset cues (e.g., /f/).

Conclusion. Results suggested that EHF hearing sensitivity facilitates rapid use of word-onset acoustic cues during speech recognition. These findings suggest that hearing sensitivity beyond the standard audiogram range (above 8 kHz) may contribute to hidden or subclinical listening differences in everyday communication and may provide useful insights into listeners' everyday listening ability beyond standard pure-tone hearing assessment.





Adapted from Lulaci et al., 2025; Lulaci, 2026.

Deep Learning–Based Classification of Hidden Cochlear Pathologies based on ABR Waterfall Plots

Prasad Darveshi (Purdue university)*; Afagh Farhadi (Purdue university); Samantha Hauser (University of Pittsburgh); Andrew Sivaprakasam (Indiana University School of Medicine); Michael Heinz (PURDUE UNIVERSITY - WEST LAFAYETTE)

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Podium

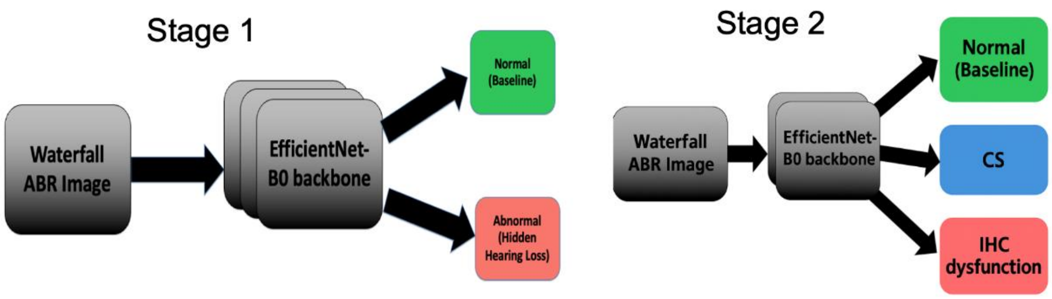
Abstract

Background. Standard Auditory Brainstem Response (ABR) interpretation often emphasizes threshold estimation, which can miss suprathreshold waveform changes associated with hidden cochlear pathology, including cochlear synaptopathy and inner-hair-cell dysfunction. We tested whether deep learning applied to ABR waterfall plots can detect morphology-based signatures of these pathologies in chinchilla models.

Methods. Chinchilla ABR datasets were analyzed from three conditions: normal-hearing baseline, noise-exposure/temporary-threshold-shift (TTS), and carboplatin-induced inner-hair-cell dysfunction. ABRs were recorded using subdermal electrodes, amplified, filtered, digitized, and averaged. For 2-, 4-, and 8-kHz stimuli, level-stacked ABR traces were converted into standardized waterfall images to preserve timing, amplitude, slope, and inter-wave morphology. We fine-tuned ImageNet-initialized EfficientNet-B0 using a two-stage pipeline. Stage 1 classified ABRs as normal or abnormal; Stage 2 classified abnormal ABRs as TTS/synaptopathy or carboplatin/IHC dysfunction. Training used augmentation, regularization, and early stopping.

Results. ABR waterfall morphology carried diagnostic information beyond threshold changes. Stage 1 showed strong detection of abnormal ABR patterns from waveform morphology alone (~91.5%). Stage 2 differentiated carboplatin-related IHC dysfunction from TTS/synaptopathy abnormalities with high performance (~83%). Most remaining errors occurred between the two abnormal groups, consistent with overlapping morphology.

Conclusions. A lightweight, morphology-aware deep-learning pipeline can identify subtle ABR waveform differences associated with hidden cochlear pathology in chinchilla models. These findings support automated ABR morphology analysis as a complement to threshold-based interpretation and motivate future work with larger datasets, optimized image construction, and translation toward individualized audiology tools for human patients.



Stage 1
Waterfall ABR Image
EfficientNet-B0 backbone
Normal (Baseline)
Abnormal (Hidden Hearing Loss)
Stage 2
Waterfall ABR Image
EfficientNet-B0 backbone
Normal (Baseline)
CS
IHC dysfunction

Investigating the Link Between Sleep Quality, Acoustic exposure, and Proxy Measures of Hidden Hearing Loss in Young Adults

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Poster

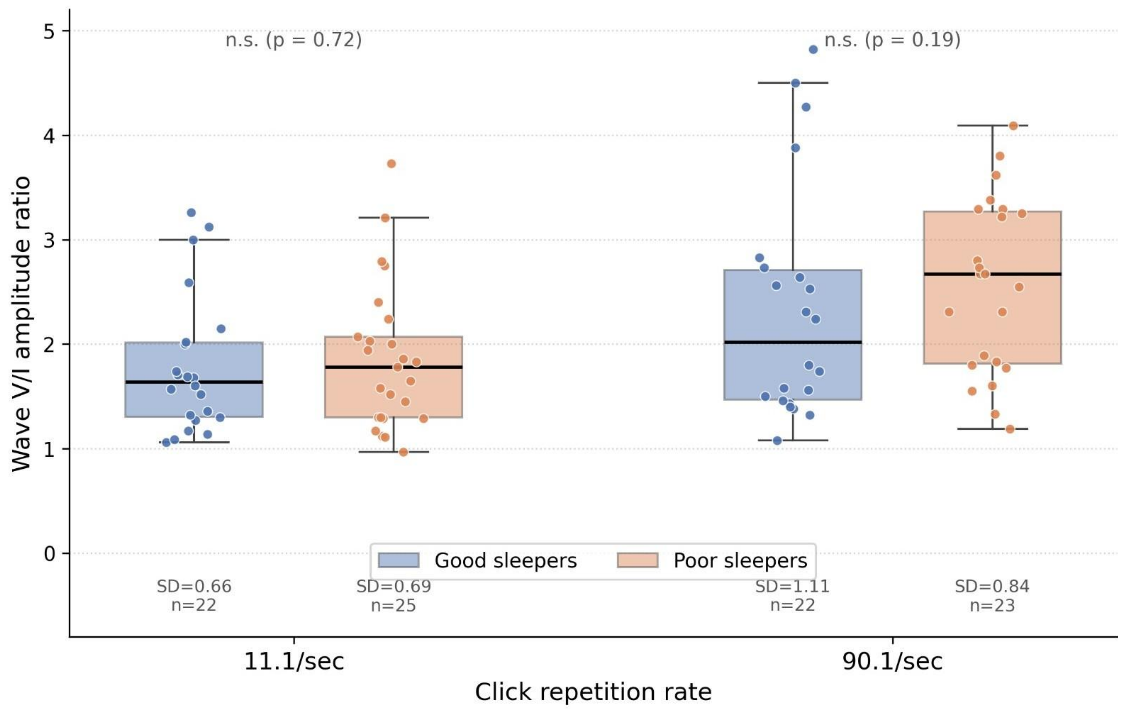
Abstract

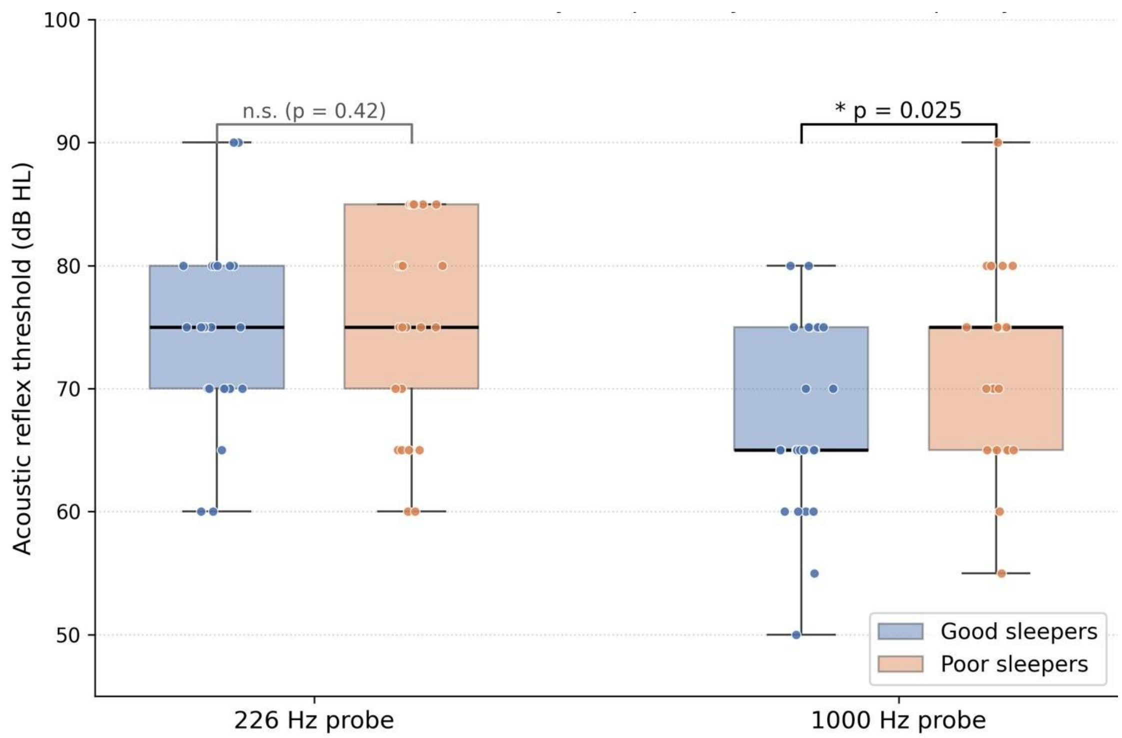
Background. While noise and music exposure are well-established risk factors for hearing health, recent evidence suggests that poor sleep quality can contribute to subtle auditory deficits.

Methods. This study investigated the association between sleep quality and proxy measures of hidden hearing loss in young adults with normal hearing sensitivity. Fifty participants were divided into two groups based on their Pittsburgh Sleep Quality Index scores: 25 with good sleep quality (≤5) and 25 with poor sleep quality (>10). A comprehensive auditory assessment was conducted, which included extended high-frequency pure tone audiometry (10–14 kHz), Amplitude modulation detection threshold test using 4000 Hz narrowband noise, Distortion Product Otoacoustic Emission input–output functions at 4000 Hz, measurement of acoustic reflex thresholds, and reflex amplitude growth using high-pass noise with 226 Hz and 1000 Hz probe tones. Additionally, auditory brainstem responses were recorded using high-pass clicks, presented at 11.1/sec and 90.1/sec. The study also explored whether acoustic exposure mediated the relationship between sleep quality and auditory function.

Results. Although the majority of auditory measures, including DPOAEs and ABR, showed no significant differences between the groups, acoustic reflex thresholds (ARTs) measured with a 1000 Hz probe tone (and not with 226 Hz probe) were significantly elevated in participants with poor sleep quality. Regression analysis revealed that sleep quality independently predicted 8.1% of the variance in ARTs, whereas acoustic exposure demonstrated minimal predictive value.

Conclusions. The preservation of cochlear mechanical function alongside elevated reflex thresholds may indicate that sleep-related auditory dysfunction primarily affects neural rather than mechanical processes. These findings provide preliminary evidence for an association between sleep quality and subtle auditory deficits.


n.s. (p = 0.72)
n.s. (p = 0.19)
Wave V/I amplitude ratio
Good sleepers
Poor sleepers
SD=0.66
n=22
SD=0.69
n=25
SD=1.11
n=22
SD=0.84
n=23
11.1/sec
90.1/sec
Click repetition rate


n.s. (p = 0.42)
* p = 0.025
Acoustic reflex threshold (dB HL)
Good sleepers
Poor sleepers
226 Hz probe
1000 Hz probe

Effect of Aging on Forward-Masking Estimates of Medial Olivocochlear Gain Reduction: Computational simulations and Behavioral Data

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Poster

Abstract

Background. In humans, aging is associated with a loss of synapses between inner hair cells and auditory nerve fibers (cochlear synaptopathy (CS)). While CS is expected to affect afferent sound encoding, it may also influence efferent control of cochlear gain by reducing the input to the medial olivocochlear (MOC) system. To date, this possibility has not been extensively investigated. Here, we combined computational simulations and psychophysics to evaluate age-related CS effects in a forward-masking task designed to assess MOC-mediated gain reduction.

Methods. We measured detection thresholds for a 2-kHz probe (10 ms) presented alone or preceded by a 400-ms noise precursor (20-ms gap) in young (yNH, n = 8) and older normal-hearing listeners (oNH, n = 7). Under these conditions, the precursor-induced threshold shift (dB) can be used to estimate cochlear gain reduction mediated by activation of the MOC system. Simulations were performed using a computational model of the auditory periphery with MOC feedback driven by inputs from the inferior colliculus and auditory nerve. Precursor bandwidth and temporal-envelope characteristics were systematically varied.

Results. Simulations predicted reduced threshold shifts in conditions simulating reduced auditory nerve input to the MOC system, particularly for narrowband precursors with weak temporal-envelope modulations. In contrast, behavioral results showed no differences between groups for narrowband precursors and significantly larger threshold shifts in oNH for wideband precursors (with and without temporal envelope modulations).

Conclusions. These results suggest that forward-masking estimates of cochlear gain reduction reflect more than MOC-mediated gain reduction alone. Future work should combine behavioral paradigms with objective measures of MOC function to better isolate age-related changes in MOC control of cochlear gain.

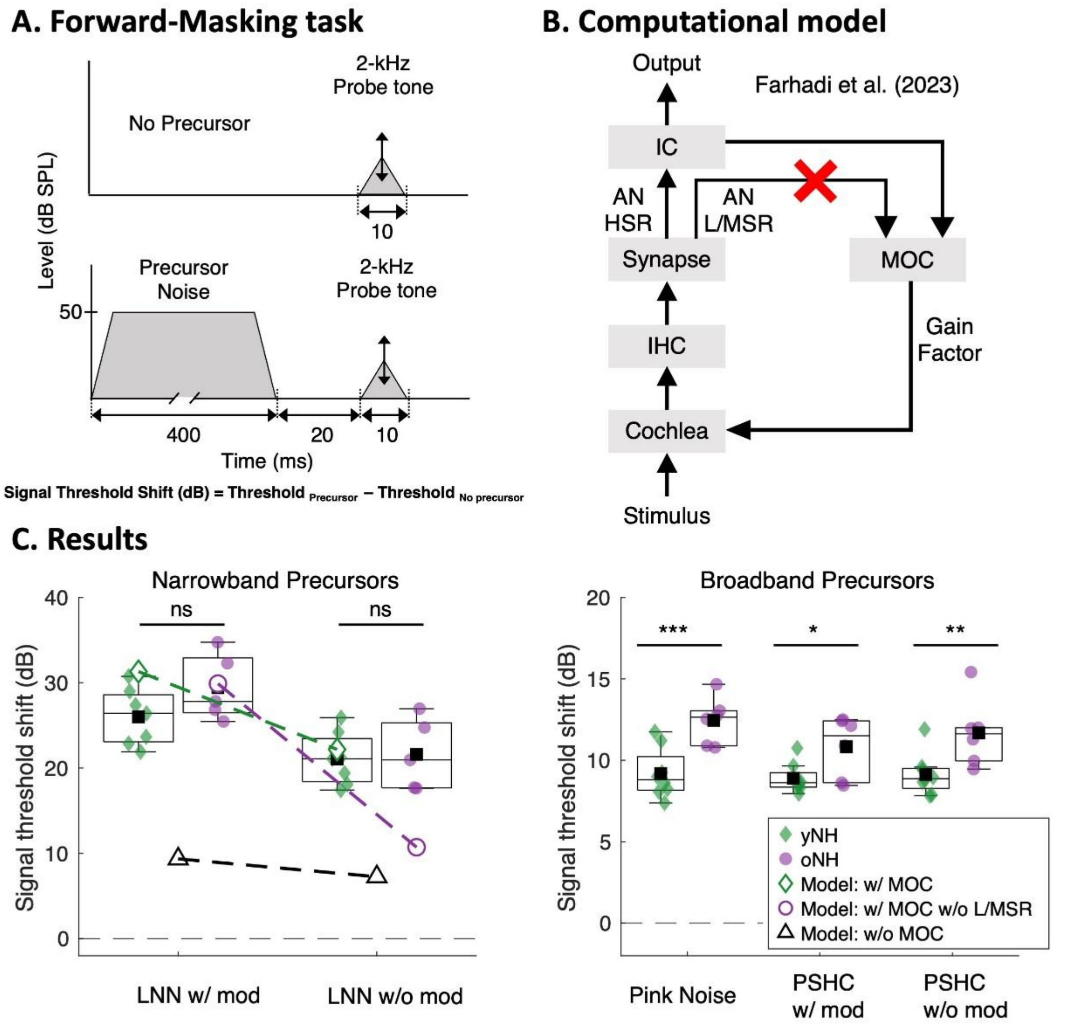

Figure 1. Summary of the study.

Panel A: Forward-masking task.

Panel B: Model used for simulations (Farhadi et al., 2023). Simulations were run with and without MOC feedback, and without AN L/MSR fibers.

Panel C: Behavioral and model results for narrowband noise precursors (right panel): LNN (low-noise noise), with and without 95-Hz temporal modulation; and behavioral results for broadband noise precursors (left panel): pink noise and PSHC (pulse-spreading harmonic complex), with and without 95-Hz temporal modulation.

Epidemiology and Profiling of Tinnitus in Adolescents

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Poster

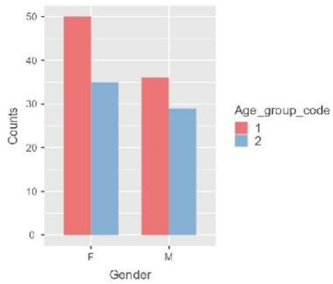
Abstract

Background. The prevalence of tinnitus among children ranges significantly from 4.7% to 46%, primarily because of variations in definitions, study methodologies, and assessment techniques. The epidemiology and clinical profiling of tinnitus in Indian adolescents attending school are still not well understood.

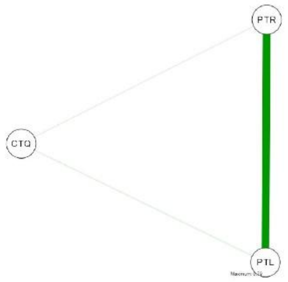
Methods. A cross-sectional study was carried out in two CBSE schools between October 2025 and January 2026. Out of 1,824 adolescents aged 10 to 17 who underwent hearing screenings, 150 individuals met the criteria for tinnitus and were assessed through detailed case histories, otoscopic examinations, pure-tone audiometry, and the Children's Tinnitus Questionnaire (CTQ). Since the data did not conform to a normal distribution, non-parametric statistical methods (Mann-Whitney U, Kruskal-Wallis, Spearman's correlation) were utilized for analysis.

Results. The prevalence of tinnitus was found to be 8.22%. All participants demonstrated normal hearing, with a mean bilateral pure tone average (PTA) of 18.4 dB HL. The CTQ score was 8.09 (SD = 4.47). The most frequently reported sound type was ringing, occurring in 28% of cases. Tinnitus mainly appeared as intermittent or occasional, with loud noises identified as the primary trigger. Those experiencing intermittent tinnitus had significantly higher CTQ scores compared to those with occasional tinnitus ($p = 0.0009$), and participants who perceived their tinnitus as loud or unpleasant also reported higher scores ($p = 0.014$). There were no notable differences based on gender, age, hearing thresholds, or duration of tinnitus.

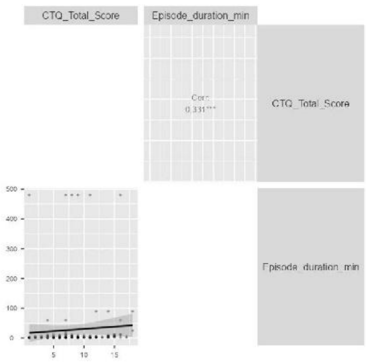
Conclusion: Tinnitus affects approximately 1 in 12 school-aged adolescents who have normal hearing. Its severity is more related to the frequency of occurrence and perceived loudness than to age, gender, or peripheral hearing in normal-hearing individuals. This emphasizes the necessity for systematic tinnitus screening within adolescent health initiatives.



Prevalence data by age and gender



Spearman’s correlation between PTA right, PTA left, and CTQ scores.



Spearman’s correlation matrix CTQ scores to tinnitus episode duration

Computational closed-loop methods to compensate for standard and hidden hearing losses

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Poster

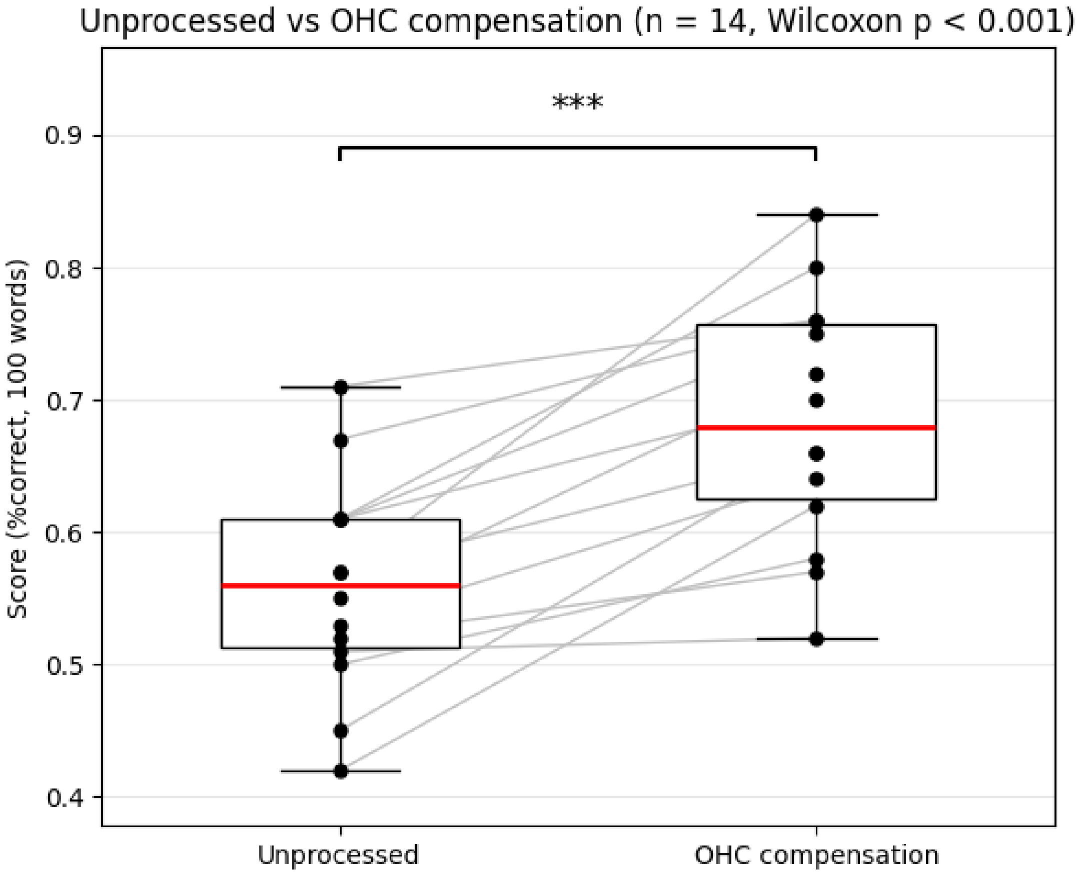
Abstract

Background. Hearing loss compensation has traditionally focused on standard audiometric deficits, where reduced audibility is addressed through amplification and frequency-specific signal shaping. However, hidden hearing loss, often associated with cochlear synaptopathy and impaired neural coding despite near-normal audiograms, can substantially degrade speech perception in noise and other real-world listening abilities. These limitations motivate computational approaches that move beyond conventional gain-based hearing aid strategies and account for both threshold and supra-threshold deficits.

Methods. We investigated computational closed-loop methods for personalized hearing compensation by combining auditory modeling and machine learning to estimate individual perceptual deficits and optimize signal processing. The evaluated approaches included speech enhancement, noise-robust feature extraction, and closed-loop fitting strategies informed by physiological measurements, with specific comparison against the NAL-NL2 prescription rule.

Results. The proposed approach showed improvements in HASPI v2 compared with NAL-NL2. A MUSHRA test with 53 patients indicated improved perceived quality over NAL-NL2 and comparable quality to the unprocessed reference condition. Preliminary performance testing of the outer hair cell (OHC) compensation algorithm in 14 patients showed an average improvement of 12.5% on a fixed-SNR 20 sentence matrix test.

Conclusions. Computational closed-loop compensation can improve personalization beyond conventional gain-based hearing aid fitting by integrating data-driven models with auditory-domain knowledge. These results support the development of next-generation assistive hearing technologies that better reflect the complexity of real hearing impairment, including both standard and hidden hearing losses.


Unprocessed vs OHC compensation (n = 14, Wilcoxon p < 0.001)
***
Score (%correct, 100 words)
0.9
0.8
0.7
0.6
0.5
0.4
Unprocessed
OHC compensation

Tuning In To the Phantom Sound

Jack Stevenson (Flinders University)*; Fauve Duquette-Laplante (Flinders University); Chengxuan (Rico) Ye (Flinders University); Deborah Katembwe (Flinders University); Brenton Hordacre (Adelaide University); Dusan Matusica (Flinders University)

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Poster

Abstract

Background. Subjective tinnitus is defined as the perception of sound in the absence of an objective external source. Here, we are investigating three neurological theories associated with the perception of tinnitus symptoms. Those being sensory gating and selective attention, where the brain fails to filter out unwanted signals within auditory networks at both pre-and-post-attentive levels respectively, and central gain, where the brain amplifies internal auditory signals, causing amplification of spontaneous neural noise that may be associated with tinnitus.

Methods. We aim to recruit 50 people, 25 with self-reported tinnitus symptoms for at least 6 months and 25 without. Our tests includes pure tone audiometry, extended high frequency and middle ear acoustic reflexes. Stroop and DigiSpan tasks will be administered to obtain a behavioural measure of attention and working memory. The Tinnitus Handicap Inventory (THI) will be used to quantify symptom severity, and other questionnaires quantifing how tinnitus affects areas such as sleep, mental health and hyperacusis will also be measured. Cortical auditory evoked potentials recordings will be performed to investigate central gain, sensory gating and attention. THI scores will be used to position participants on a scale of high-severity tinnitus to low-severity/no tinnitus and differences in the EEG and behavioural measures will be compared across this scale.

Results. Data collection and analysis is currently ongoing, with 8 participants (3 with tinnitus and 5 without) having completed the study to date.

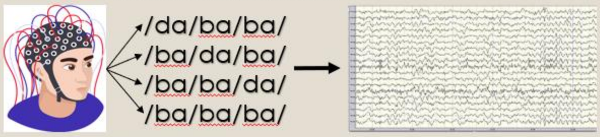
Conclusions. This study hopes to identify any potential differences across individuals with varying levels of tinnitus severity. We hypothesize that we will see more reduction in both the P50 and P300, while seeing increases in the P2 evoked potentials as THI scores increase across individuals.

Tuning In To the Phantom Sound

What are we investigating?

We are investigating brain activity using EEG to determine the impact of tinnitus, focusing on three main theories:

Selective Attention: /da/ and /ba/ presented at different probabilities in +10dB SNR



Sensory Gating: 40ms /da/ presented in a pair

Central Gain: 158ms long /da/ presented in noise at 3 different signal to noise ratios

Speech-in-Noise as a Biomarker of Peripheral Auditory Neuropathy in Metabolic Disorders: Proof of Concept in Patients with Uncontrolled Type 2 Diabetes

Camille Rochmann (CILCARE SAS); Zoe Kaier-Green (CILCARE Inc); Blandine Duval (CILCARE SAS); Hugo Laullier (CILCARE SAS); Laura Breda (CILCARE SAS); Mathieu Schué (CILCARE SAS)*

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Poster

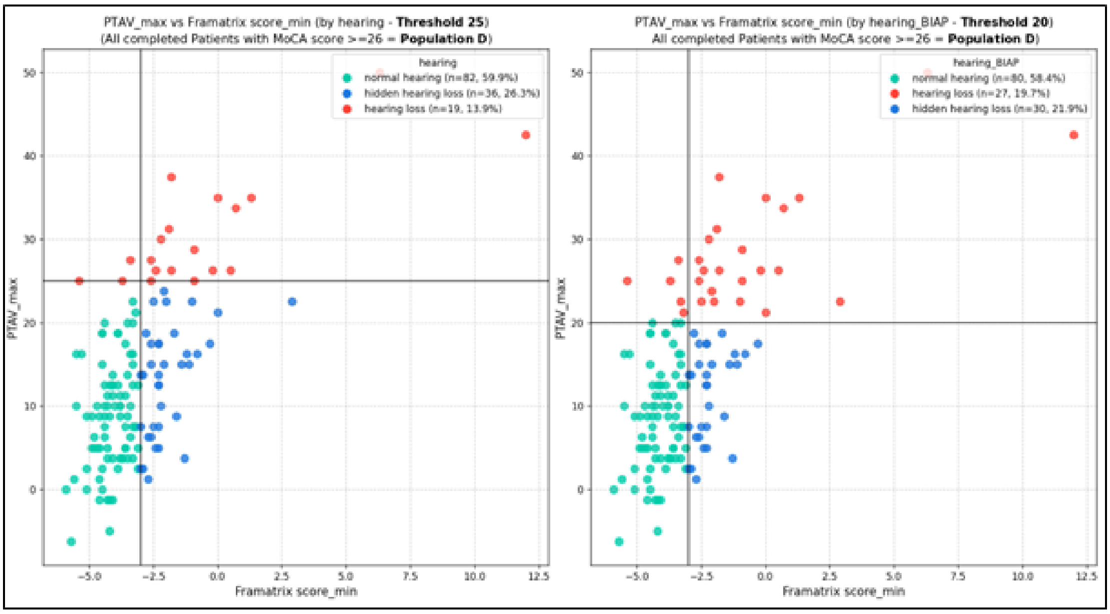
Abstract

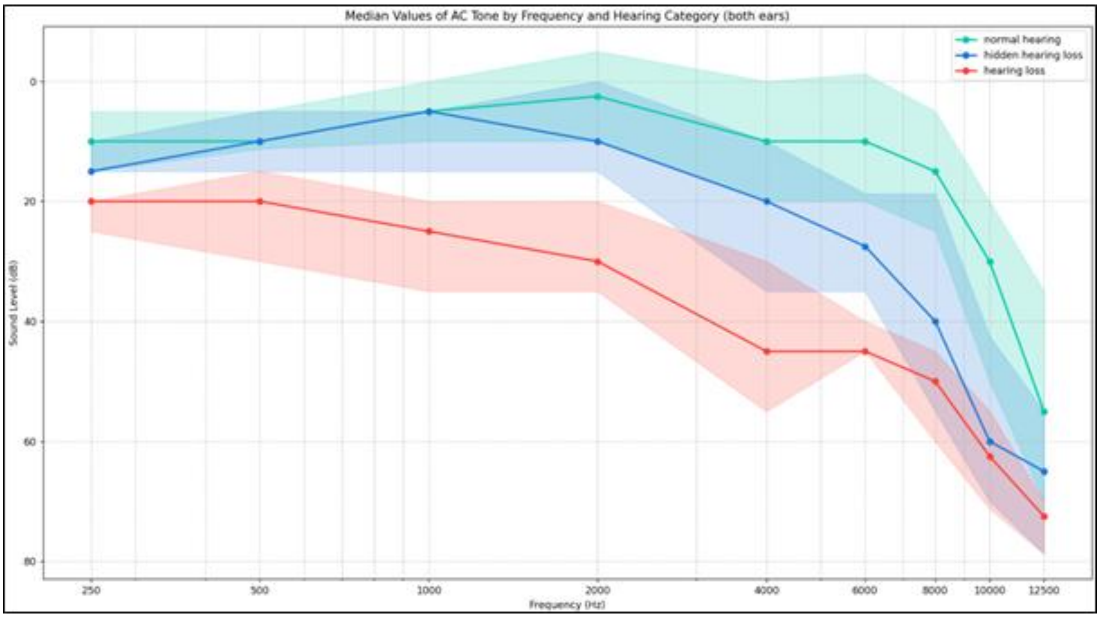
Background. Sensorineural hearing loss related to diabetes is increasingly documented, with possible causes including cochlear microangiopathy, chronic inflammation, or peripheral auditory neuropathy. These factors may contribute to speech-in-noise intelligibility deficit despite normal standard audiograms, often reported as “hidden hearing loss”.

Methods. DIAMANT was a French, multicenter, cross-sectional, prospective study conducted in France to evaluate the prevalence of various auditory dysfunctions in uncontrolled type-2 diabetes (T2D) patients. Participants who consented (aged 25–70, T2D >2 years, HbA1c >7%) underwent a comprehensive battery of tests: tonal (0.25-16 kHz) audiometry, vocal audiometry in silence and in noise (French Matrix test (Framatrix)), Distortion Product OtoAcoustic Emissions, electrocochleography (EcochG), and cognitive screening via the Montreal Cognitive Assessment (MoCA). Participants with abnormal otoscopy, tympanometry, conductive / asymmetrical hearing loss were excluded.

Results. Among 241 participants (61% women, median age 60 [IQR:53.0-64.0]), three hearing profiles emerged.After excluding those with cognitive impairment (MoCA <26), 137 participants remained: 59.8% Normal hearing, 26.3% SIN-D, and 13.9% HL. NH and SIN-D had similar thresholds at low frequencies (0.25–1 kHz), but SIN-D diverged at mid-to-high frequencies (≥2 kHz), resembling HL at 8-16 kHz. At group level, EcochG revealed significantly reduced auditory nerve response to suprathreshold stimulations in SIN-D compared to NH, reflecting a decrease or degeneration of synapses and/or spiral ganglion cells.

Conclusion. Although SNHL prevalence in T2D was not higher than in the general population, SIN deficit with normal hearing was more frequent. SIN testing offers a practical, non-invasive alternative to electrophysiology and should be integrated into routine auditory testing. More work should confirm SIN testing as a biomarker of peripheral auditory neuropathy.


PTAV_max vs Framatrix score_min (by hearing - Threshold 25)
(All completed Patients with MoCA score >=26 = Population D)
hearing
normal hearing (n=82, 59.9%)
hidden hearing loss (n=36, 26.3%)
hearing loss (n=19, 13.9%)
PTAV_max
Framatrix score_min
PTAV_max vs Framatrix score_min (by hearing_BIAP - Threshold 20)
All completed Patients with MoCA score >=26 = Population D)
hearing_BIAP
normal hearing (n=80, 58.4%)
hearing loss (n=27, 19.7%)
hidden hearing loss (n=30, 21.9%)
PTAV_max
Framatrix score_min


Median Values of AC Tone by Frequency and Hearing Category (both ears)
normal hearing
hidden hearing loss
hearing loss
Sound Level (dB)
Frequency (Hz)

Session 3.B. Objective Measures and Biomarkers in Hearing Research

Electrophysiological, pupillometric, and behavioural indices for assessment and prognosis.

Chaired by Dr. Andrew (Andy) J. Beynon.

Multichannel Auditory Cortical Evoked Potentials and Source Localization Analyses in Misophonia: A Neurophysiological Investigation

Kamalakannan Karupaiah (All India Institute of Speech and Hearing, Mysore)*; Ajith Kumar Uppunda (Professor of Audiology, Department of Audiology and Centre for Hering Sciences, All India Institute of Speech and Hearing, Mysuru, India.); Prashanth Prabhu (Assistant Professor, Department of Audiology,All India Institute of Speech and Hearing, Mysuru, India.)

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Featured talk

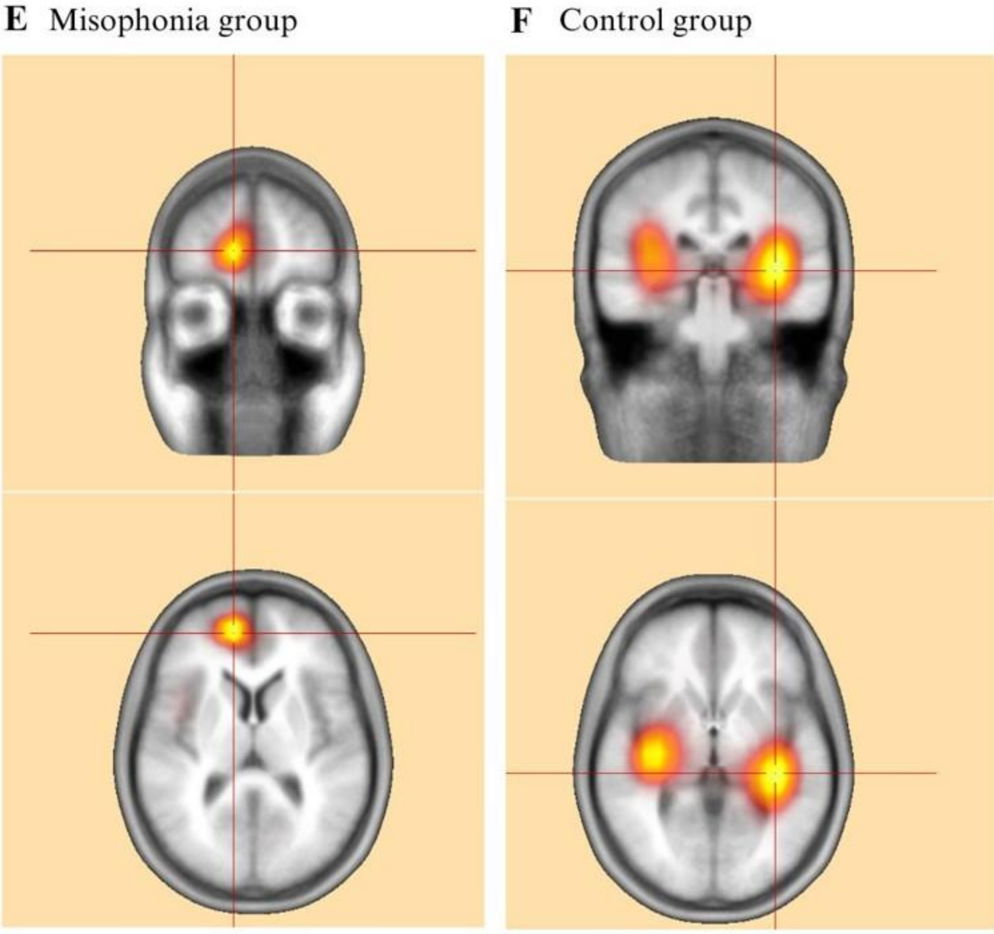
Abstract

Background. Misophonia involves heightened emotional and physiological responses to specific auditory triggers, but its neurophysiological basis remains unclear. Cortical auditory evoked potentials (CAEPs) findings in misophonia are mixed. Multichannel CAEP with source localization analyses could identify subtle neural alterations and the underlying cortical generators. This study aimed to investigate auditory cortical function and delineate neural source activity associated with misophonia using auditory late latency responses (ALLRs).

Methods. Eighty adults (18–30 years old) with normal hearing were divided into the misophonia (n=40) and control (n=40) groups. Individuals with Misophonia were identified using established inclusion criteria and the MisoQuest questionnaire. ALLRs were recorded using a 32-channel EEG system, and peak latencies, amplitudes, scalp topography, and source activity were analyzed.

Results. Individuals with misophonia exhibited significantly reduced ALLR amplitudes and earlier peak latencies at Fz, Cz, and Pz ($p < .05$). Scalp maps showed a shift from fronto-central activation (controls) to centro-parietal dominance (misophonia). Source analysis revealed increased right medial frontal gyrus activation in misophonia and greater left temporal lobe activation in controls.

Conclusions. Misophonia is associated with atypical auditory cortical processing, altered temporal dynamics, and increased frontal involvement. Distinct neural activation patterns identified through source localization highlight potential biomarkers of misophonia and provide insight into its underlying pathophysiology, with implications for the development of targeted therapeutic interventions.



Talaraich coordinates:
X = 11.3, Y= 47.3, Z = 9.7
Right Cerebrum,
Medial Frontal Gyrus

Talaraich coordinates:
X = -35.1, Y= -28.7, Z = 2.1
Left Cerebrum,
Temporal Lobe

Measurement-Aware Registry Design for Cochlear Implant Outcomes: Lessons from a 25-Year Clinical Dataset

Apurv Shukla (University of Michigan)*; Amy Paoletti (University of Michigan Medicine); Devin McCaslin (University of Michigan Medicine); Pravansu Mohanty (University of Michigan Dearborn)

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Podium

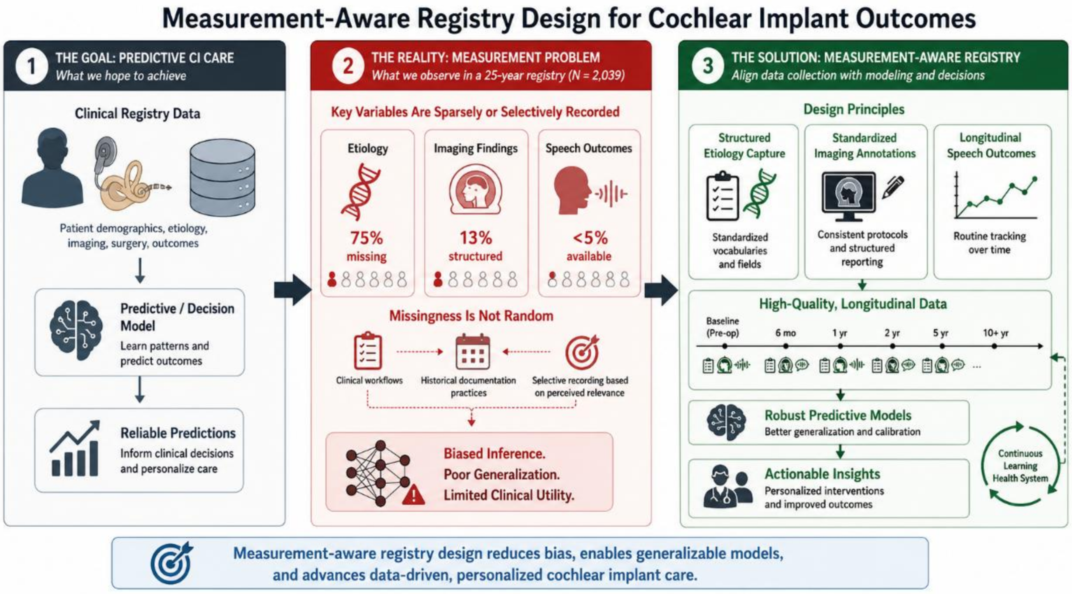
Abstract

Advances in computational audiology increasingly rely on real-world clinical data to develop predictive and decision-support models for cochlear implant (CI) care. However, the structure and completeness of existing clinical registries remain poorly aligned with these goals. In particular, key variables—such as etiology, imaging findings, and longitudinal speech outcomes—are often inconsistently documented, introducing systematic measurement bias that limits downstream modeling.

We analyze a 25-year single-center cochlear implant registry comprising 2,039 adult patients to characterize the data-generating and measurement processes underlying CI records. While clinically meaningful structure is evident—etiology stratifies patients by age, onset type, imaging findings, and adverse outcomes—we find that critical variables are sparsely or selectively recorded. Etiology is undocumented in 75% of patients, imaging findings are structured in only 13%, and speech outcomes are available for fewer than 5%, despite their central role in evaluating CI success .

These missingness patterns are not random: they reflect clinical workflows, historical documentation practices, and selective recording based on perceived relevance. As a result, naïve analyses risk biased inference, and standard machine learning pipelines are unlikely to generalize.

We argue for a measurement-aware registry design paradigm in computational audiology, where data collection is explicitly aligned with modeling and decision-making objectives. Specifically, we highlight the need for (i) structured etiology capture, (ii) standardized imaging annotations, and (iii) longitudinal speech outcome tracking. Our findings provide both an empirical characterization of current limitations and a concrete roadmap for designing CI registries that support predictive modeling and personalized intervention.


Measurement-Aware Registry Design for Cochlear Implant Outcomes
1 THE GOAL: PREDICTIVE CI CARE
What we hope to achieve
Clinical Registry Data
Patient demographics, etiology, imaging, surgery, outcomes
Predictive / Decision Model
Learn patterns and predict outcomes
Reliable Predictions
Inform clinical decisions and personalize care
2 THE REALITY: MEASUREMENT PROBLEM
What we observe in a 25-year registry (N = 2,039)
Key Variables Are Sparsely or Selectively Recorded
Etiology
75% missing
Imaging Findings
13% structured
Speech Outcomes
<5% available
Missingness Is Not Random
Clinical workflows
Historical documentation practices
Selective recording based on perceived relevance
Biased Inference.
Poor Generalization.
Limited Clinical Utility.
3 THE SOLUTION: MEASUREMENT-AWARE REGISTRY
Align data collection with modeling and decisions
Design Principles
Structured Etiology Capture
Standardized vocabularies and fields
Standardized Imaging Annotations
Consistent protocols and structured reporting
Longitudinal Speech Outcomes
Routine tracking over time
High-Quality, Longitudinal Data
Baseline (Pre-op)
6 mo
1 yr
2 yr
5 yr
10+ yr
Robust Predictive Models
Better generalization and calibration
Actionable Insights
Personalized interventions and improved outcomes
Continuous Learning Health System
Measurement-aware registry design reduces bias, enables generalizable models, and advances data-driven, personalized cochlear implant care.

Triggered: using computer vision to measure sound intolerance

Samuel Smith (Mass Eye and Ear, Harvard Medical School)*; Jenna Sugai (Mass Eye and Ear); Nicole Buie (Mass Eye and Ear, Harvard Medical School); Kenneth Hancock (Mass Eye and Ear, Harvard Medical School); Daniel Polley (Mass Eye and Ear, Harvard Medical School)

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Podium

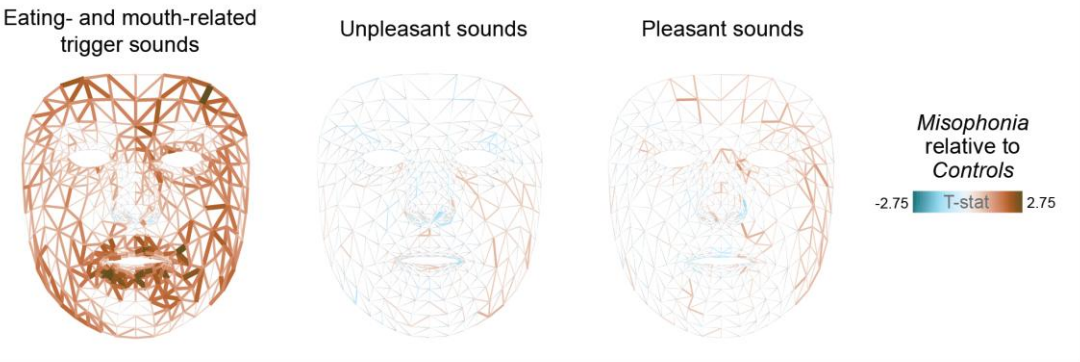
Abstract

Background. Unlike senses such as smell, it has long been thought that "hearing isn't a strong vector for disgust" (Aurel Kolnai, 1929). Yet, millions of people with misophonia would disagree. These are individuals who are triggered, disgusted and distressed by eating- and mouth-related sounds. A socially debilitating disorder, the experience of misophonia has no established measures beyond self-report.

Methods. The clinical burden of misophonia appears closely linked to affective reactivity. We hypothesized this may manifest as elevations in physiological and autonomic measures. A cohort of 61 participants (34 misophonia, 27 controls) listened to a diverse battery of sounds, including common misophonia triggers such as eating- and mouth-related sounds. Simultaneously, quantitative facial videography measurements were made, alongside pupillometric tracking and heart rate recordings. Statistical analyses included group and individual-level differences, and interactions with sound-type and level.

Results. We observed robust evidence that eating- and mouth-related sounds trigger distinct autonomic and physiological responses in individuals with misophonia. Quantitative videography revealed significant elevations in lower face movements, accompanied by increased pupil dilations and heart rates. Differences between misophonia and control groups were specific to sound-type rather than sound-level. Preliminary modeling suggests these combined metrics provide a viable signature for discriminating between misophonia and control groups.

Conclusions. These results highlight that eating- and mouth-related triggers cause substantial distress in individuals with misophonia, accompanied by measurable elevations in autonomic and physiological responses. Current efforts focus on refining these metrics into a predictive model capable of classifying clinical status and quantifying the degree of distress using contact-free technology.


Eating- and mouth-related trigger sounds
Unpleasant sounds
Pleasant sounds
Misophonia relative to Controls
-2.75
T-stat
2.75

Altered Central Auditory Gain in Misophonia: N1–P2 Slope and Categorical Loudness Findings

Ishita Marwaha (All India Institute of Speech and Hearing (AIISH), Mysore)*; Prashanth Prabhu (All India Institute of Speech and Hearing (AIISH), Mysore)

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Podium

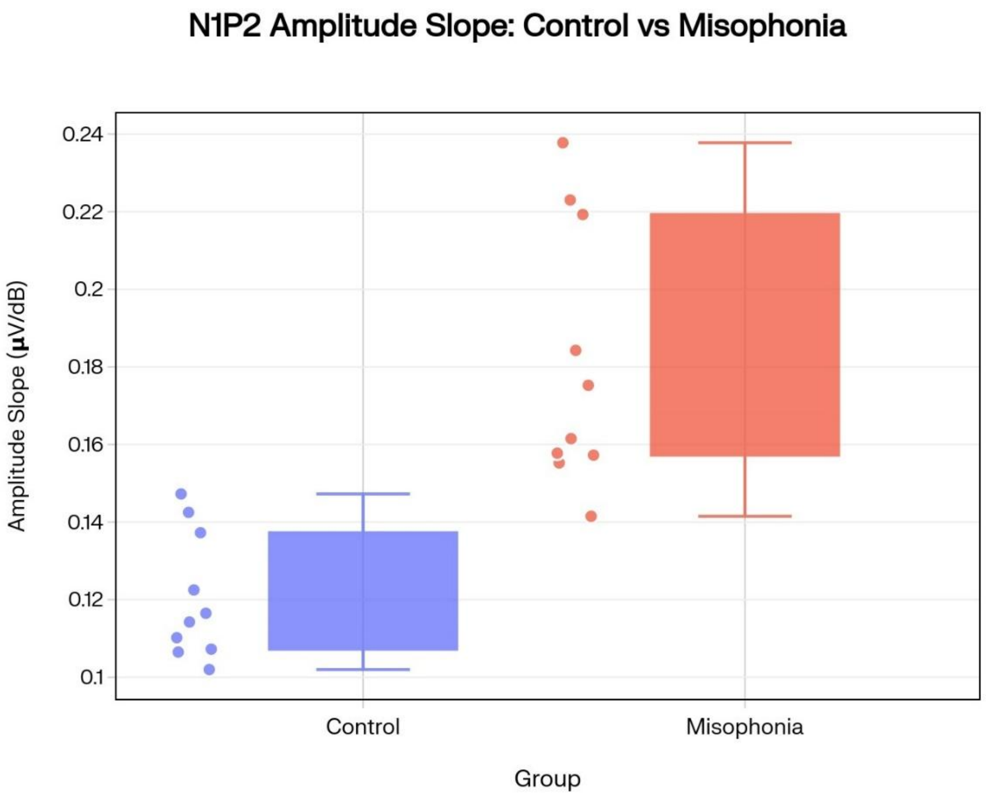
Abstract

Background. Misophonia involves extreme aversive reactions to specific sounds, potentially reflecting altered central auditory gain. The loudness dependence of auditory evoked potentials (LDAEP) (N1–P2 amplitude slope), serves as an established biomarker of cortical serotonergic function and intensity-dependent loudness processing, but remains unexplored in the context of misophonia. We hypothesized steeper N1–P2 slopes and altered subjective loudness scaling, indicating enhanced cortical gain and compressed dynamic range.

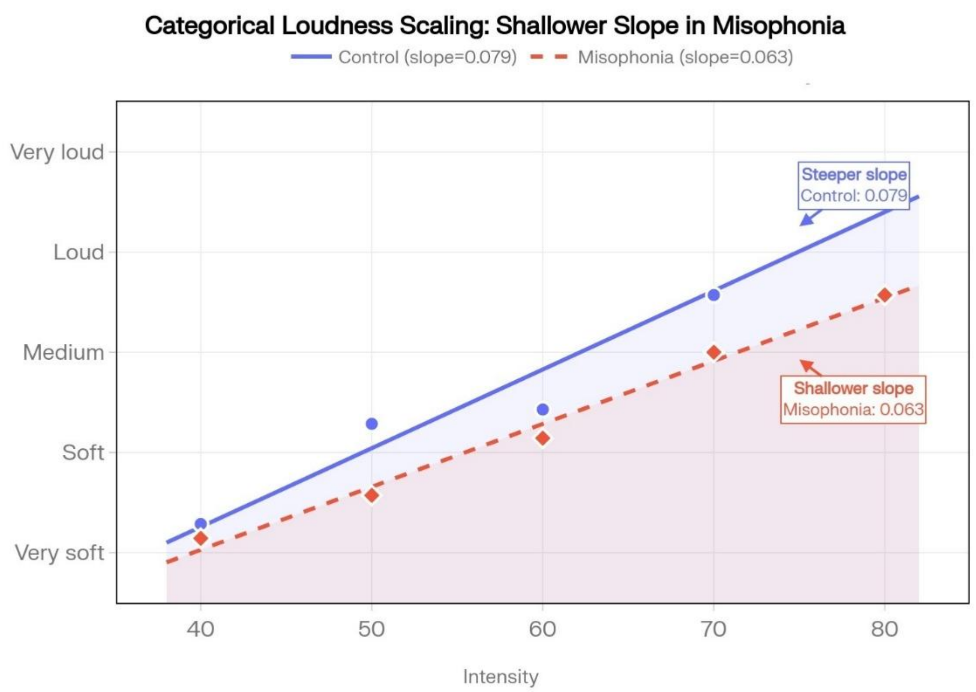
Methods. 20 adults (10 controls, 10 misophonia; normal hearing). LDAEP with 1 kHz tones (40-80 dB nHL; 80ms: 10ms rise-fall; 0.5/s; 150 sweeps) via ER-3A inserts. Vertical montage: Cz/non-inverting, M1&M2/inverting, Fpz/ground (1-30 Hz bandpass, 50k gain). Subjects watched captioned video (eyes open). Primary outcome: N1–P2 amplitude slopes. Categorical loudness scaling using 7-point scale (ISO 2006: Extremely Loud→ Not Heard). Welch's independent t-tests compared groups (α=0.05). Post-hoc power analysis evaluated effect robustness (observed Cohen's d values and unequal variance assumptions).

Results. Misophonia demonstrated significantly steeper N1–P2 slopes (M=0.184 µV/dB, SD=0.035) vs controls (M=0.129 µV/dB, SD=0.023; $t(7.3)=3.29$, $p=0.008$, $d=1.79$). Categorical loudness scaling revealed trend towards shallower slopes in misophonia (M=0.072, SD=0.012) vs controls (M=0.085, SD=0.008; $t=2.14$, $p=0.07$, $d=0.68$). Post-hoc power exceeded 78% for N1–P2 large effect ($d=1.79$) and 65% for loudness medium-large effect despite modest sample, confirming observed differences exceed chance expectation.

Conclusions. Misophonia manifests enhanced objective cortical gain alongside compressed subjective loudness rating, suggesting dissociated central auditory processing with gain-compression mismatch. This finding positions N1–P2 LDAEP slope as promising objective biomarker for misophonia diagnosis, though larger replication studies needed to confirm robustness.



N1P2 Amplitude Slope: Control vs Misophonia
Amplitude Slope (µV/dB)
0.24
0.22
0.2
0.18
0.16
0.14
0.12
0.1
Control
Misophonia
Group



Categorical Loudness Scaling: Shallower Slope in Misophonia
Control (slope=0.079)
Misophonia (slope=0.063)
Very loud
Loud
Medium
Soft
Very soft
Steeper slope
Control: 0.079
Shallower slope
Misophonia: 0.063
40
50
60
70
80
Intensity

Thalamocortical Auditory Processing in Misophonia: Evidence From Middle Latency Auditory Evoked Potentials

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santhoshperiyasamycbe@gmail.com

Podium

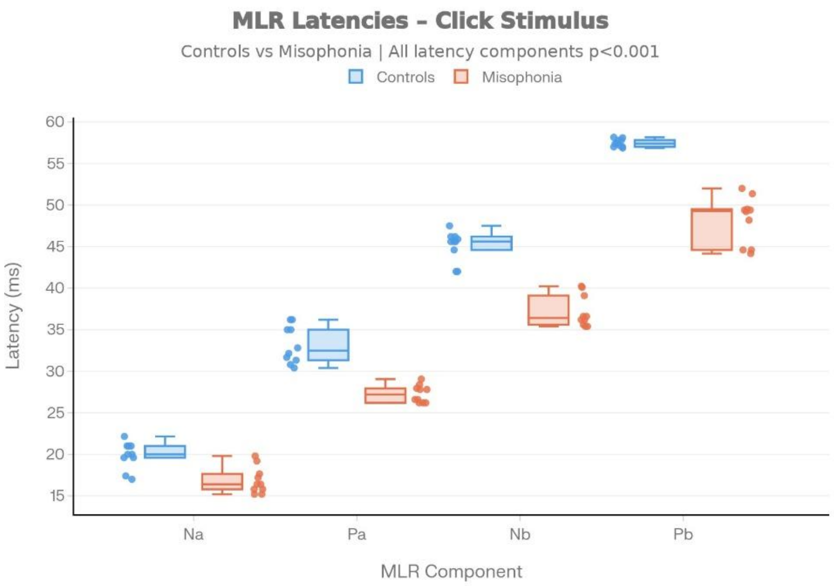
Abstract

Background. Misophonia is a sound tolerance disorder characterised by intense emotional and autonomic responses to specific pattern sounds. Despite growing interest in its neurophysiological underpinnings, objective evidence from the middle latency response (MLR), a marker of thalamocortical auditory transmission remains unexplored. This study investigated MLR latency and amplitude characteristics in individuals with misophonia across multiple stimulus conditions.

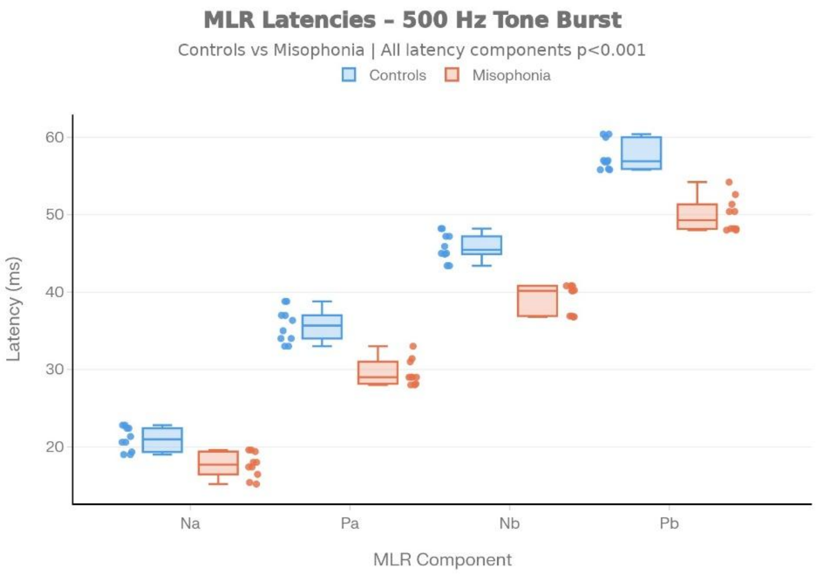
Methods. Twenty adults participated: 10 with clinically identified misophonia and 10 age- and gender-matched controls. MLR was recorded using click, 500 Hz tone burst, and 4 kHz tone burst stimuli. Latencies and amplitudes of Na, Pa, Nb, and Pb components were compared using independent samples t-tests with Cohen's d effect sizes.

Results. MLR latency components were significantly shorter in the misophonia group across all stimulus conditions ($p < 0.001$). Effect sizes were uniformly large for click (Na: $t=4.18$, $d=1.87$; Pa: $t=7.52$, $d=3.36$; Nb: $t=9.61$, $d=4.30$; Pb: $t=10.16$, $d=4.55$), 500 Hz (Na: $t=4.78$, $d=2.14$; Pa: $t=7.00$, $d=3.13$; Nb: $t=8.23$, $d=3.68$; Pb: $t=8.25$, $d=3.69$), and 4 kHz tone bursts (Na: $t=4.99$, $d=2.23$; Pa: $t=6.94$, $d=3.11$; Nb: $t=5.69$, $d=2.55$; Pb: $t=6.94$, $d=3.11$). Amplitudes showed no significant between-group differences under any condition, indicating that neural timing rather than neural magnitude is the primary neurophysiological distinguishing feature in misophonia.

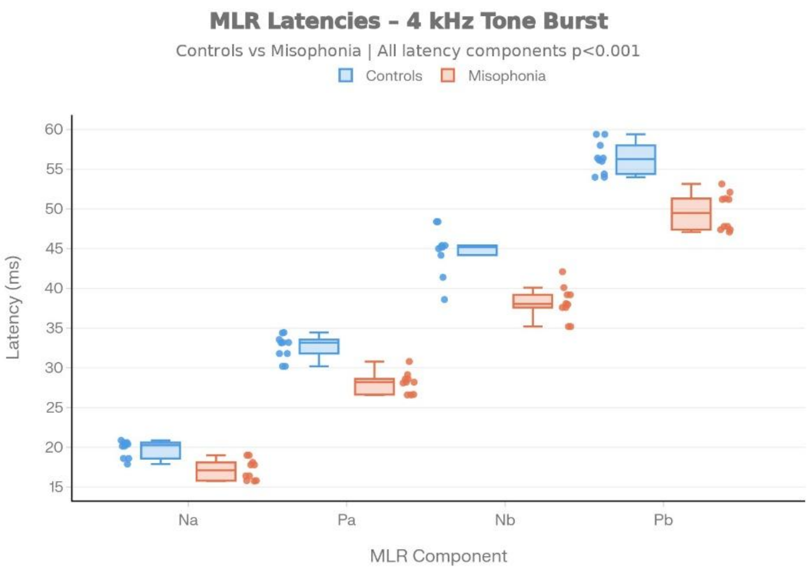
Conclusions. Misophonia is characterised by significantly shorter MLR latencies with preserved amplitudes, indicative of accelerated but not augmented thalamocortical transmission. This dissociation between latency and amplitude points to a timing-based neural hyperexcitability and supports MLR as a promising objective tool in misophonia assessment.



MLR Latencies - Click Stimulus
Controls vs Misophonia | All latency components p<0.001
Controls
Misophonia
Latency (ms)
60
55
50
45
40
35
30
25
20
15
Na
Pa
Nb
Pb
MLR Component



MLR Latencies - 500 Hz Tone Burst
Controls vs Misophonia | All latency components p<0.001
Controls
Misophonia
Latency (ms)
60
50
40
30
20
Na
Pa
Nb
Pb
MLR Component



MLR Latencies - 4 kHz Tone Burst
Controls vs Misophonia | All latency components p<0.001
Controls
Misophonia
Latency (ms)
60
55
50
45
40
35
30
25
20
15
Na
Pa
Nb
Pb
MLR Component

Otoacoustic Emission Screening Extended to the Assessment of Auditory Neural Health

Francois Deloche (Macquarie University)*; Sanna Hou (National Acoustic Laboratories); Matthew Croteau (National Acoustic Laboratories); Thomas Fung (Macquarie University); Georgy Sofronov (Macquarie University); Viji Easwar (National Acoustic Laboratories); Sriram Boothalingam (Macquarie University)

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Podium

Abstract

Background. Newborn Hearing screening programs commonly use auditory brainstem responses (ABR) or otoacoustic emissions (OAEs) to detect hearing loss. A current advantage of ABRs over OAEs is that they can detect neural deficits like auditory neuropathy spectrum disorders (ANSD). We have developed a technique that repurposes the click stimulus and evoked emissions to monitor brainstem reflexes, potentially detecting and aiding in the differential diagnosis of ANSD with just OAEs.

Methods. Repeated click trains of 1s were used to evoke OAEs (80 and 90 dB ppSPL, presentation rate: 64 Hz). Responses in the [0–4 ms] and [6–15 ms] windows following each click were analysed to monitor the middle ear muscle (MEMR) and medial olivocochlear (MOCR) reflexes, respectively. In normal-hearing subjects, these reflexes produce a change in the response amplitude over the course of the stimulus. Reflex detection was based on a likelihood-ratio score assuming a two-term exponential time course under normal hearing (H1), and a time-invariant model under sensory hearing loss or ANSD (H0).

Results. OAEs were recorded bilaterally in a group of 64 children; 24 were normal hearing, 20 had sensorineural loss, 15 had conductive loss, and 5 had ANSD. Hearing loss detection was evaluated using receiver operating characteristic (ROC) analysis and compared with a baseline method based on OAE signal-to-noise ratio (SNR). The best performance was obtained by combining MEMR and MOCR detection (areas under curve: OAE SNR 0.87, MOCR 0.89, MEMR 0.9, MOCR+MEMR 0.93). Ongoing work examines whether spectral cues (magnitude and phase) can further improve the specificity of reflex detection.

Conclusion. OAEs alone can be used to screen for cochlear and neural deficits. Our approach, requiring only conventional OAE recording hardware and fewer operational resources, offers a cost-effective alternative for newborn and post-newborn hearing screening and diagnostics.

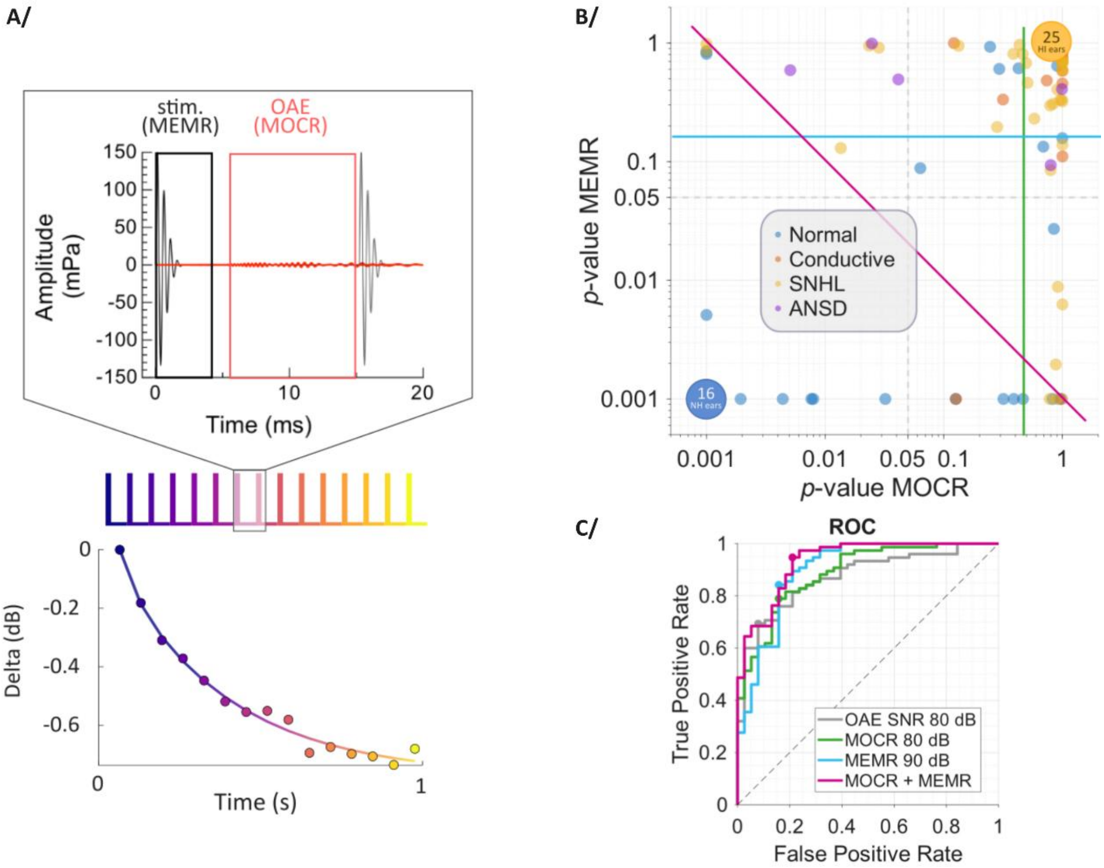

Figure: **A/** Stimulus paradigm. Top: Response to one stimulus block with the analysis windows. Center: Click train. Bottom: Time-course of the efferent modulation of the response (in this example, MOCR).
B/ Detection of the MEMR and MOCR (p-values) for 64 subjects × 2 lateralities, separated into normal hearing and hearing impaired groups (see color code). **C/** Detection of hearing loss: performance evaluated with a ROC curve, based on the reflex p-values or OAE SNR. MOCR+MEMR: combined via Fisher's method. The optimal points (circle) maximize Youden's score and correspond to the thresholds shown in B/ with same colors.

Exploring Auditory Encoding in Misophonia Using Frequency Following Response

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Poster

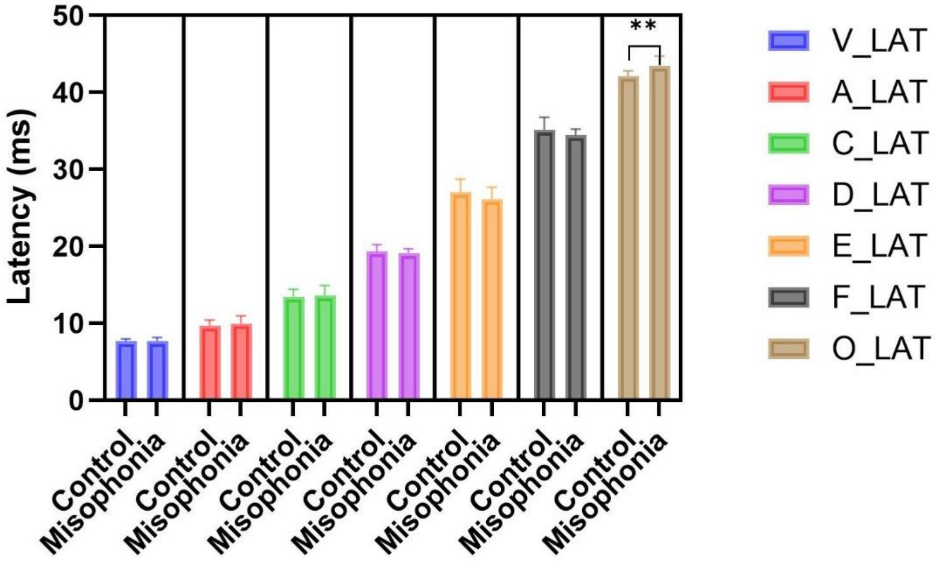
Abstract

Background. Misophonia is characterized by strong aversive emotional reactions to specific sounds. Frequency Following Response (FFR) provides insights into the subcortical neural encoding of sound, with contributions from brainstem structures, which may help explain the heightened reactivity to everyday auditory stimuli reported in misophonia. The present study aimed to investigate auditory encoding objectively using FFR in individuals with misophonia.

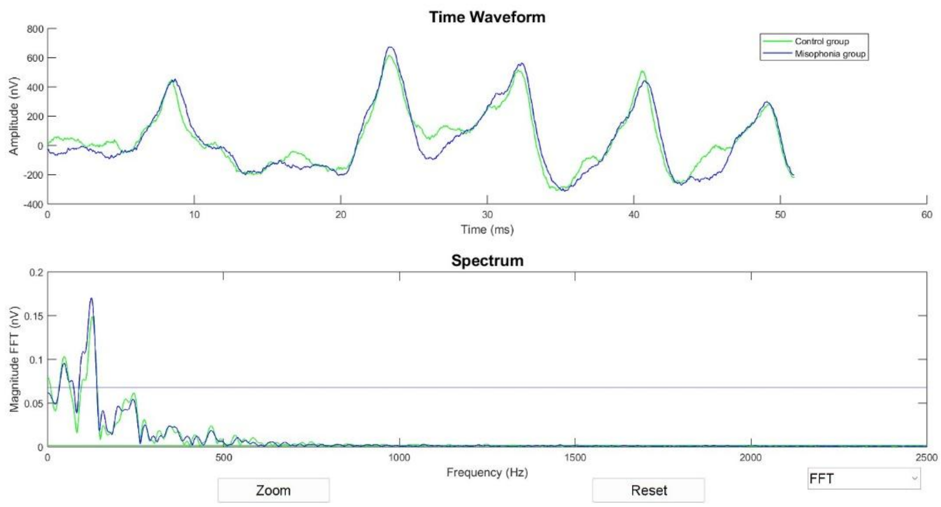
Methods. Thirty participants were recruited and divided equally into two groups: a misophonia group (n = 15) and a control group (n = 15). Individuals with misophonia were identified using Schröder's diagnostic criteria (2014), and severity was assessed using the Revised Amsterdam Misophonia Scale (AMISOS-R). FFR was recorded using the Intelligent Hearing Systems (IHS) SmartEP with a 40 ms /da/ stimulus, presented at 70 dB SPL, with a repetition rate of 4.1/s, via ER-3 insert earphones, binaurally. Offline analysis included identifying FFR peaks and extracting latency and amplitude values.

Results. The independent-samples t-test revealed a significant difference in the latency of peak O between the two groups. The offset peak was significantly delayed ($p = 0.001$) in the misophonia group compared to the control group. These findings are consistent with previous work by Kim et al. (2023), who reported a delay in ABR Wave V latency in individuals with misophonia, collectively suggesting delayed auditory neural processing at the brainstem level.

Conclusions. The results suggest a possibility of delayed subcortical auditory encoding of speech in individuals with misophonia. This aberrant neural timing at the brainstem level may reflect an underlying auditory processing anomaly that predisposes individuals with misophonia to disproportionate emotional and physiological responses to specific sounds.



Latency (ms)
50
40
30
20
10
0
**
Control
Misophonia
V_LAT
A_LAT
C_LAT
D_LAT
E_LAT
F_LAT
O_LAT



Time Waveform
Control group
Misophonia group
Amplitude (nV)
Time (ms)
Spectrum
Magnitude FFT (nV)
Frequency (Hz)
Zoom
Reset
FFT

Auditory Sensory Gating in Misophonia: A Neurophysiological Evaluation

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Poster

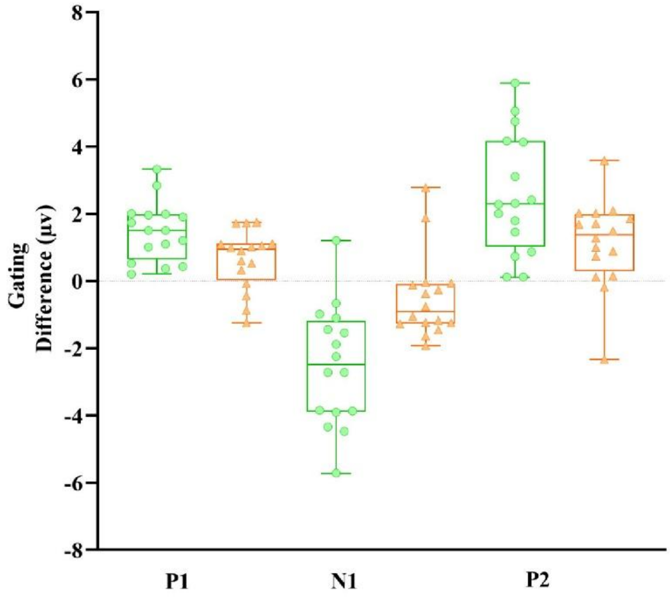
Abstract

Background. Misophonia is characterized by strong emotional and physiological reactions to specific everyday sounds such as chewing or breathing. While behavioral symptoms are well documented, the underlying neural mechanisms remain unclear. Auditory sensory gating, a pre-attentive inhibitory process that filters redundant auditory information, provides an objective measure of cortical inhibitory function. This study aimed to examine auditory sensory gating in individuals with and without misophonia using cortical auditory evoked potentials (CAEPs).

Methods. Thirty adults aged 18–35 years participated, including 15 individuals with misophonia and 15 age- and gender-matched controls. Inclusion was based on Schröder et al. (2013) criteria and the MisoQuest Questionnaire (Siepsiak et al., 2020). Sensory gating was assessed using a conditioning–testing (S1–S2) paradigm with paired 250 Hz tone bursts (50 ms duration, 500 ms interstimulus interval, 7 s interpair interval). CAEPs were recorded at the Cz electrode site using a four-channel Intelligent Hearing Systems SmartEP module. P1, N1, and P2 peak latencies and amplitudes were analyzed for S1 and S2, and gating ratios (S2/S1 × 100) were calculated. Associations between gating indices and Revised Amsterdam Misophonia Scale (RAMISO-S) scores were also evaluated.

Results. Individuals with misophonia showed significantly reduced N1–P2 amplitude suppression ($p < .05$) compared to controls, indicating attenuated sensory gating and diminished cortical inhibition. Higher RAMISO-S scores were associated with poorer gating efficiency, suggesting that greater symptom severity relates to impaired auditory regulation.

Conclusions. These findings demonstrate distinct neurophysiological alterations in misophonia, characterized by impaired auditory sensory gating. Such deficits may serve as potential electrophysiological markers of abnormal cortical inhibitory processing and support integrating electrophysiological measures into diagnostic and therapeutic frameworks.



8
6
4
2
0
-2
-4
-6
-8
Gating
Difference (µv)
P1
N1
P2

Inference of Cochlear Amplifier Gain Using Simulated Distortion-Product Otoacoustic Emission Level Maps

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Poster

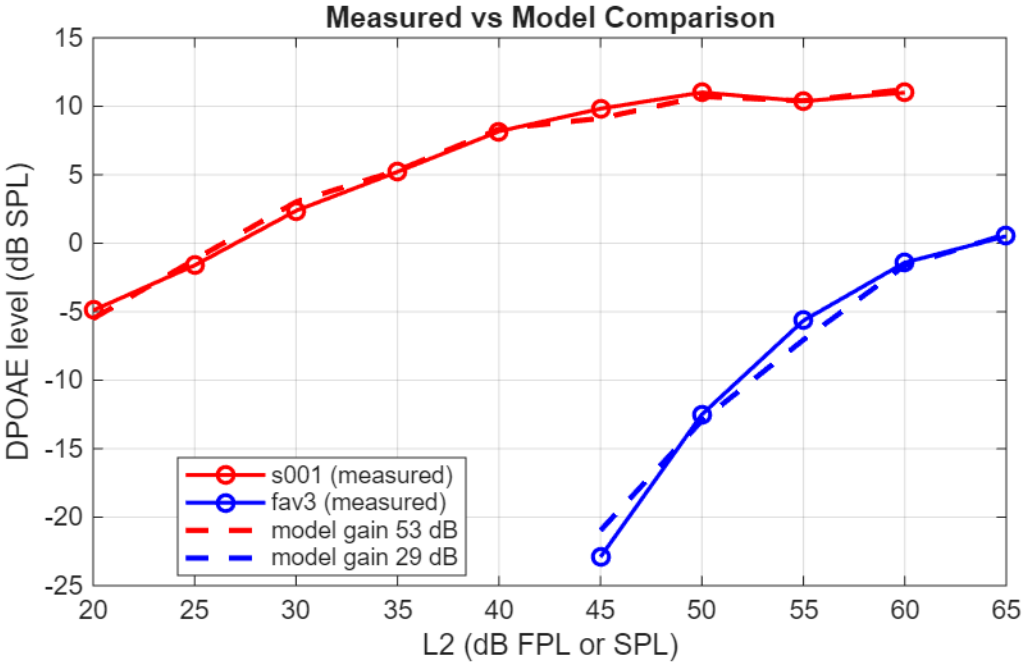
Abstract

Background. Distortion-product otoacoustic emissions (DPOAE) growth functions, i.e., DPOAE amplitudes as a function of stimulus level, can be used to estimate the DPOAE threshold, which correlates with the absolute hearing threshold. However, the shape and level of DPOAE growth functions are not only affected by cochlear amplifier gain, but also by middle-ear forward and reverse transmission. Therefore, the clinical use of DPOAE growth functions is still very limited.

Methods. A cochlear model is used to predict DPOAEs for various combinations of stimulus levels. DPOAE amplitude shown as a function of the level is called the DPOAE level map. DPOAE level maps derived from the cochlear model with uniformly set amplification at frequencies from 1 kHz to 6 kHz can be aligned, indicating the distortion invariance of the cochlear model. DPOAE growth functions are measured for the "scissor paradigm" and fitted into the simulated DPOAE level maps for various gains of cochlear amplification to determine cochlear amplifier gain.

Results. DPOAE growth functions are very sensitive to changes in cochlear amplifier gain. Even changes within the normal audiometry range (20 dB) yield strong changes in the shape of DPOAE growth functions, as shown for subject s001 at 2 kHz with a hearing threshold of 5 dB HL and subject fav3 with a hearing threshold of 20 dB HL at 2 kHz. The growth function of subject s001 is very well fitted into DPOAE level maps derived from the model with a gain of 53 dB, and the growth function of subject fav3 corresponds to a model gain of 29 dB.

Conclusion: The proposed method of alignment with simulated DPOAE provides a time efficient framework allowing for the measurement of the DPOAE growth function without the necessity to determine the most optimal stimulus levels yielding the largest DPOAE amplitudes. This technique is therefore suitable in combination with swept stimuli, allowing for the separation of distortion and reflection components of DPOAEs.



Measured vs Model Comparison
DPOAE level (dB SPL)
L2 (dB FPL or SPL)
s001 (measured)
fav3 (measured)
model gain 53 dB
model gain 29 dB
15
10
5
0
-5
-10
-15
-20
-25
20
25
30
35
40
45
50
55
60
65

Age-Related Alteration of Neural Correlates of Tone-in-Noise Detection in the Auditory Midbrain

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Poster

Abstract

Background. Age-related hearing loss is a progressive sensorineural hearing deterioration that is one of the three most prevalent chronic medical conditions of our elderly, impacting one-third of the global population over 65 years old – largely impacting one's ability to hear in background noise. The loss of peripheral inputs and senescence-related alterations of central neurotransmission lead to decreased activity driving neurons in the inferior colliculus (IC), a major midbrain convergence site critical for processing complex sounds such as speech. However, how aging affects hearing in noisy environments at the level of neural encoding within the IC remains largely unresolved. To investigate this, we measured signal-in-noise detection in the IC of young and old CBA/CaJ mice.

Methods. We used in vivo extra-cellular electrophysiology to measure the responses of neuronal populations in the in the central nucleus of the IC of 11 young and 15 old CBA/CaJ mice while listening to tones-in-noise. Tone detection was quantified via a d' sensitivity index and characterized as either excitatory or inhibitory and temporally categorized into onset or sustained responses.

Results. Overall, there were 4.7% more onset responses in the old group compared to the young group (53.1% vs. 48.4%). Additionally, there were 6.2% more onset responses relative to sustained within the old group (53.1% vs. 46.9%) compared to 3.2% less onset responses relative to sustained in the young group (48.4% vs. 51.6%). Young mice had an increased proportion of tone detection and detection strength between both response patterns across most frequency and SNR conditions. However, we found that units in old mice which were robust tone detectors showed larger inhibitory-sustained responses to tones-in-noise.

Conclusions. Our findings suggest that the aging IC contains distinct subpopulations of neurons that preferentially encode different signal-in-noise features and exhibit altered response properties.

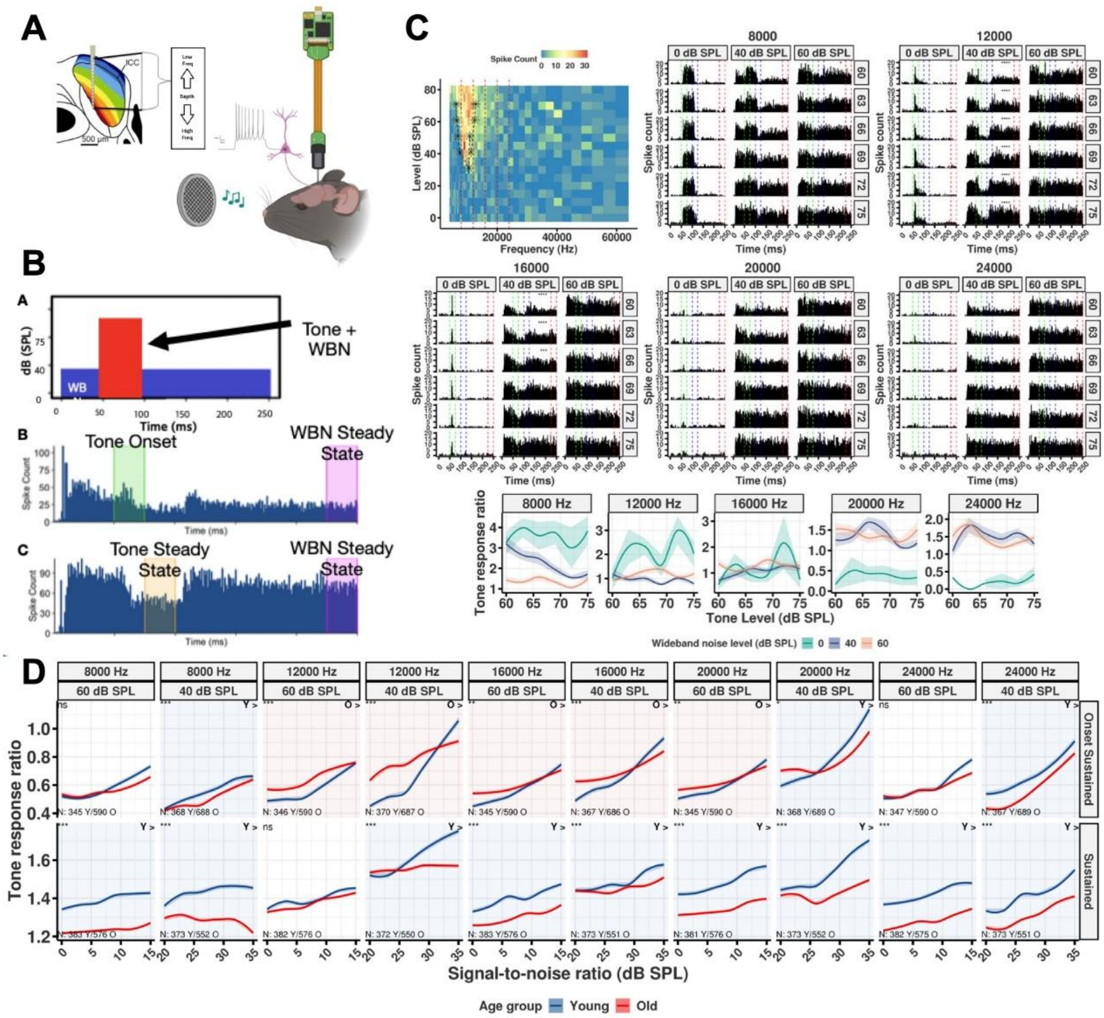

Fig. 1. Recording setup, tone-in-noise stimulus, and auditory midbrain response properties. (A) *In vivo* recording in the inferior colliculus. **(B)** Tone-in-noise stimulus presentation diagram. **(C)** Old unit with a sharp low-frequency receptive field that demonstrates alternating response properties based on the frequency of the tone. At 8 kHz, a strong sustained-adapting response to the tone-in-quiet/noise is present. However, at 12 and 16 kHz, the dynamic shifts to an excitatory onset response for the tone-in-quiet but inhibitory sustained response for the tone-in-noise. At 20 and 24 kHz, the unit is unresponsive to the tone. **(D)** For each frequency, noise condition, and wideband noise response type, a grand average tone response ratio value was computed (counts in tone onset/steady-state window). Detection strength generally increased with SNR, along with the young group demonstrating stronger detection for sustained responses. For onset sustained responses, the old group had higher initial detection ratios for lower SNRs at 12-20 kHz. As SNR increased, the young group gradually progressed to maintain a higher detection strength at high SNRs.

Multi-Response Deconvolution preserves pABR Estimation Quality Despite Increasing Response Overlap

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Poster

Abstract

Background. The parallel auditory brainstem response (pABR) enables simultaneous acquisition of frequency-specific ABRs across ears, substantially reducing recording time compared to serial paradigms. In pABR, responses are estimated by averaging EEG segments associated with Poisson-distributed stimuli (conventional pABR averaging). However, simultaneous stimulation introduces overlapping responses that may degrade estimation quality.

Methods. A simulation framework was developed to characterise the effect of increasing the number of simultaneous stimuli on pABR estimation quality. Noise-free EEG signals were synthesised as convolutions of Poisson-distributed event sequences with up to 10 different ABR templates. Responses were estimated using conventional pABR averaging and multi-response deconvolution. Monte Carlo simulations evaluated response quality in terms of signal-to-noise ratio (SNR).

Results. Figure 1 shows that, with conventional pABR averaging, SNR progressively decreases as the number of simultaneous stimuli increases, whereas deconvolution maintains stable SNR across conditions. Figure 2 shows that, despite the SNR degradation observed with conventional pABR averaging, the resulting effect on ABR morphology is negligible due to the large temporal jitter of Poisson-distributed sequences. Figure 3 shows that deconvolution preserves both SNR and waveform morphology independently of the number of overlapping responses.

Conclusions. Increasing the number of simultaneous stimuli in the pABR paradigm reduces estimation SNR when responses are obtained through conventional pABR averaging. However, this degradation has minimal practical impact on ABR morphology. Multi-response deconvolution minimises the SNR decrease, preserving response quality even in highly overlapping conditions. These findings support deconvolution as a robust strategy for pABR analysis in paradigms involving large numbers of simultaneous stimuli.

Figure 1. Effect of parallel stimuli on pABR quality

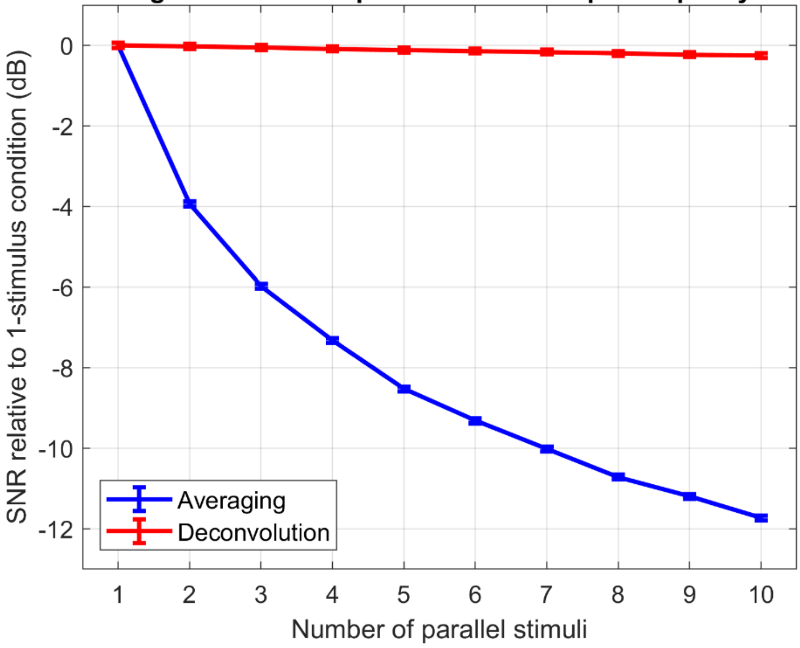


Figure 2. pABR quality and waveform estimates using AVERAGING

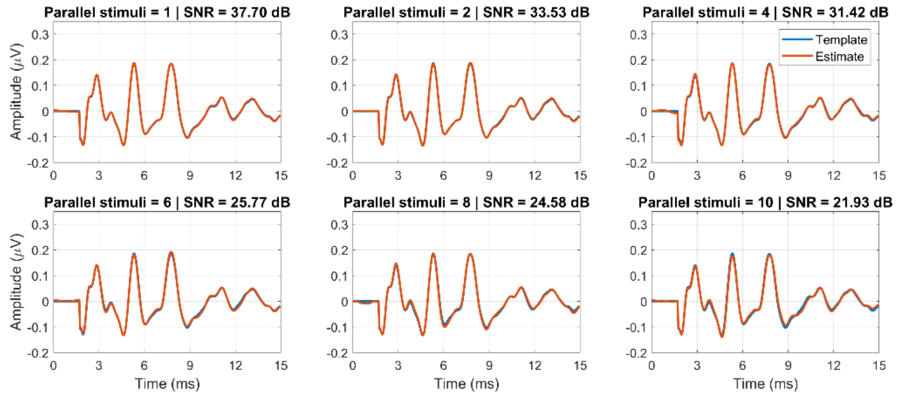
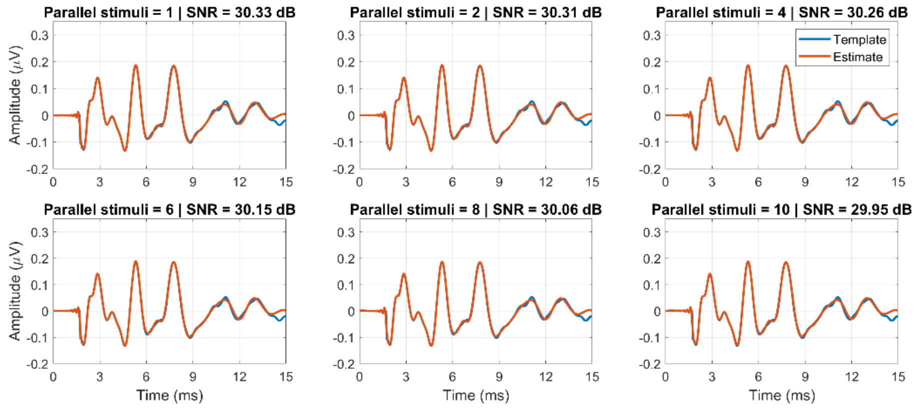


Figure 3. pABR quality and waveform estimates using DECONVOLUTION



Effect of Context on Cortical Speech Tracking in Young Adults with Normal-Hearing

Fauve Duquette-Laplante (Flinders University)*; Cyril Alex (Flinders University); Shivani Prabhu (Flinders University); Varghese Peter (University of the Sunshine Coast); Mridula Sharma (Flinders University)

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Poster

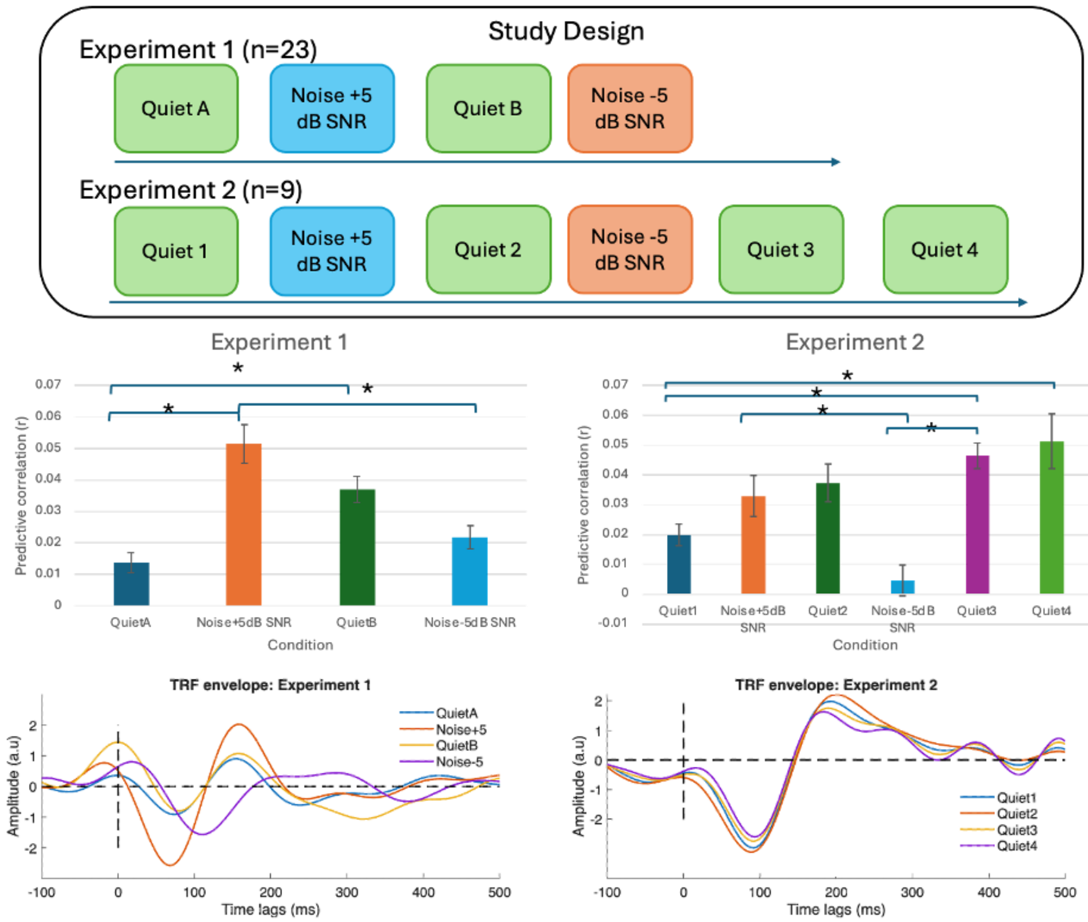
Abstract

Background. Cortical tracking of the speech envelope, measured using multivariate temporal response function (mTRF) analysis, has been used to investigate neural speech processing in noise. Low levels of background noise have been associated with enhanced neural responses, potentially reflecting stochastic resonance. However, repeated exposure to speech may also influence cortical tracking through attentional or learning effects. This study examined the impact of noise on envelope features and the priming effect of noise on listening in quiet in adults with normal hearing.

Methods. In Experiment 1(EXP1), 23 young adults with normal hearing listened to continuous speech presented in four conditions: QuietA, +5 dB signal-to-noise ratio (SNR) (+5), QuietB, and −5 dB SNR (-5). To investigate the priming effect of noise on quiet, Experiment 2 (EXP2) was conducted with 9 other participants, using the sequence: Quiet2, +5, Quiet2, -5, Quiet3, and Quiet4. Speech was presented in freefield at 00 azimuth, 1.3 meters from the participant at 60dB SPL. EEG data were analyzed using mTRF envelope reconstruction, and predictive correlations were compared across conditions.

Results. Impact of noise: EXP1 demonstrated significantly higher predictive correlations for +5 than for QuietA ($p<.001$) and -5 ($p<.001$), and EXP2 demonstrated a greater predictive correlation for Noise+5 than for Noise-5 ($p=.007$). Priming effect: in EXP1, QuietA had a lower predictive correlation than QuietB ($p<.001$). EXP2 showed that Quiet3 ($p=.002$) and Quiet4 ($p=.039$) had significantly stronger predictive correlations than Quiet1.

Conclusions. Although enhanced neural tracking in moderate noise (+5) may be consistent with stochastic resonance theories, the present findings suggest that priming effects or statistical learning are causing the preceding noise to affect subsequent speech perception. This highlights the importance of presentation order and exposure effects when designing EEG paradigms.



Study Design
Experiment 1 (n=23)
Quiet A
Noise +5 dB SNR
Quiet B
Noise -5 dB SNR
Experiment 2 (n=9)
Quiet 1
Noise +5 dB SNR
Quiet 2
Noise -5 dB SNR
Quiet 3
Quiet 4
Experiment 1
Predictive correlation (r)
QuietA
Noise+5dB SNR
QuietB
Noise-5dB SNR
Condition
Experiment 2
Quiet1
Noise+5dB SNR
Quiet2
Noise-5dB SNR
Quiet3
Quiet4
TRF envelope: Experiment 1
Amplitude (a.u)
Time lags (ms)
QuietA
Noise+5
QuietB
Noise-5
TRF envelope: Experiment 2
Quiet1
Quiet2
Quiet3
Quiet4

Test–Retest Reliability and Agreement of Ear-Canal and Mastoid Wave I Auditory Brainstem Responses in Healthy Young Adults

Blandine Duval (CILCARE SAS)*; Laura Breda (CILCARE SAS); Charles Vincent (CILCARE SAS); Zoe Kaier-Green (CILCARE Inc); Jérôme Geoffroy (CILCARE SAS); Mathieu Schué (CILCARE SAS); Hugo Laullier (CILCARE SAS)

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Poster

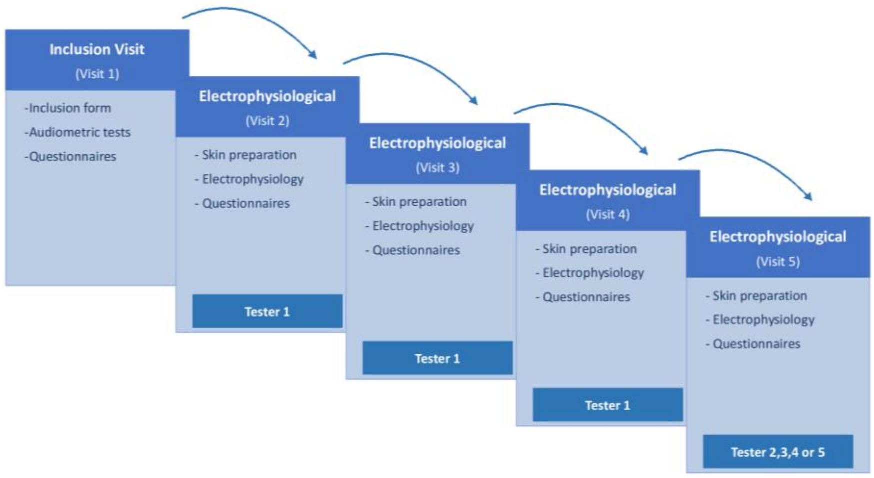
Abstract

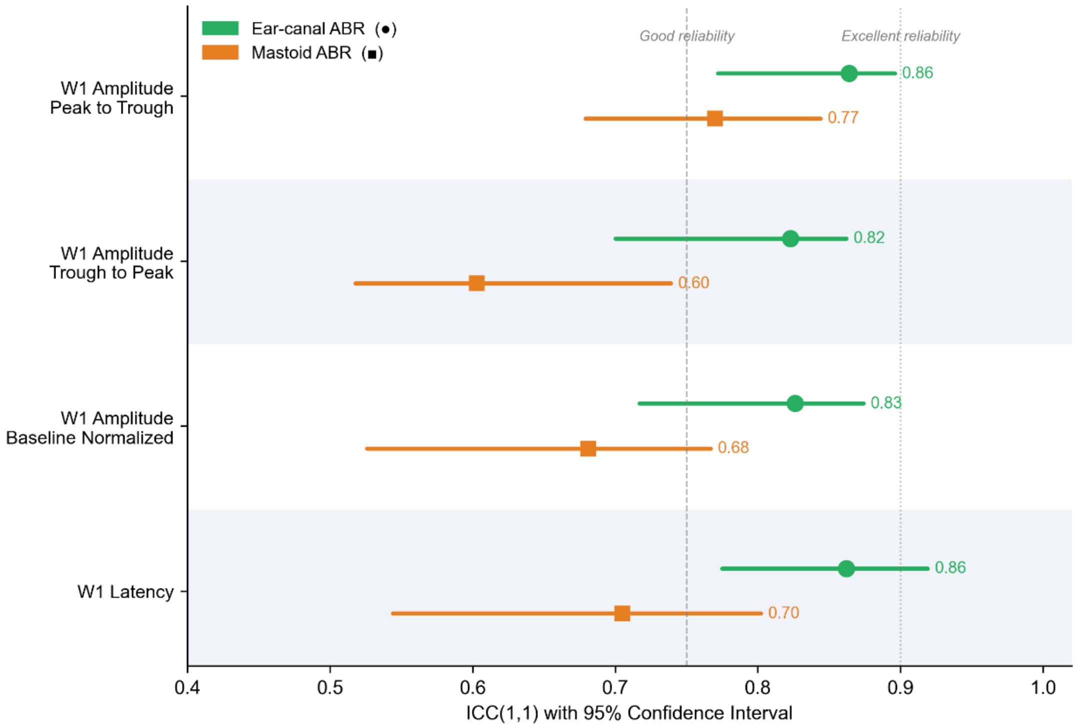
Background. Wave I amplitude is a proposed non-invasive biomarker of cochlear synaptopathy, but established reproducibility is required before clinical use.

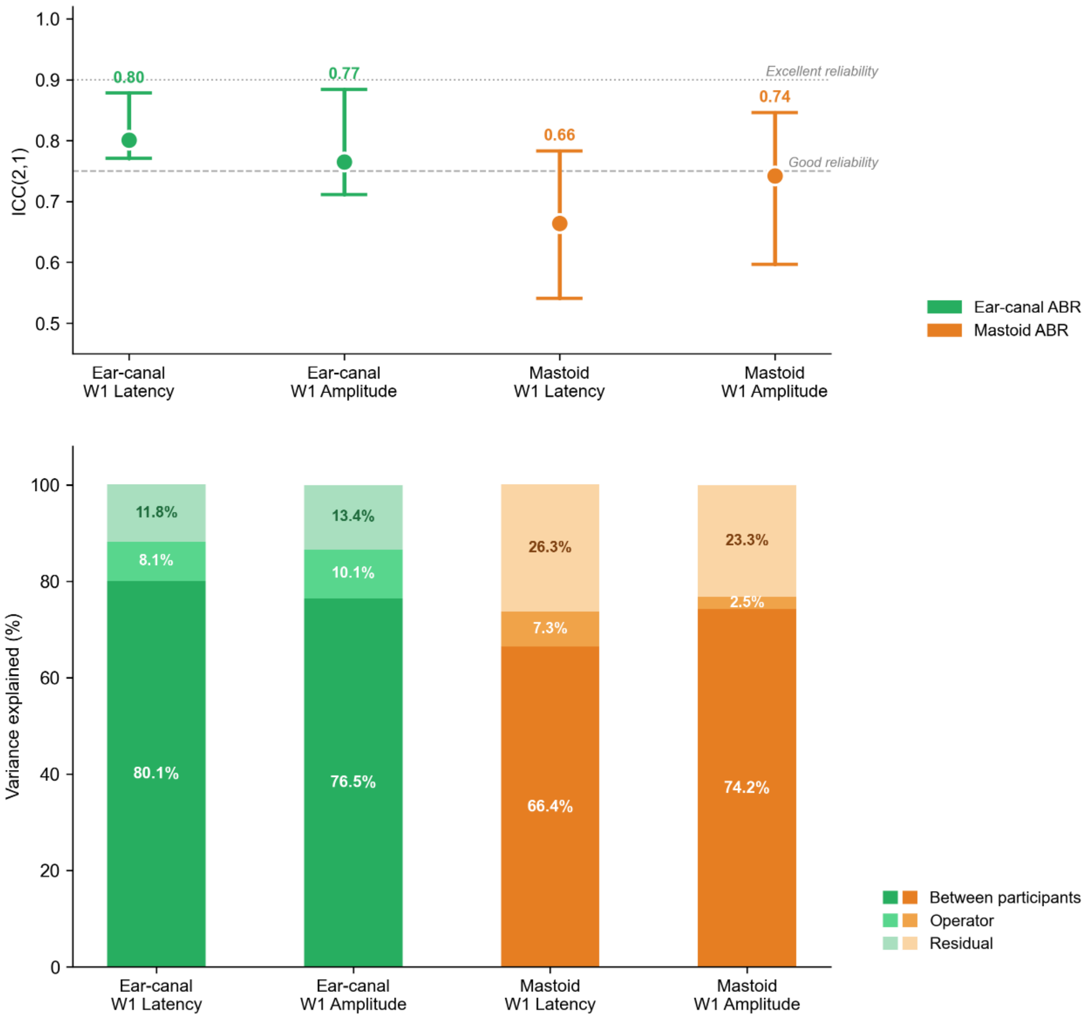
Methods. Thirty healthy adults aged 18–25 years (19 women) with normal audiometric thresholds (Pure-Tone Average 0.5–4 kHz < 20 dB HL) and normal speech-in-noise performance (SNR < −3 dB) attended up to four electrophysiological sessions two weeks apart. Click-evoked ABRs were recorded at 80 dB nHL with ECG monitoring: three sessions by the same tester, one by another tester. Test–retest reliability of Wave I amplitude and latency was evaluated, and sources of measurement variability, including head size, cardiac and respiratory parameters, and tester effects, were examined.

Results. An SNR-based quality threshold retained ~90% of recordings. Reproducibility was higher for Ear-canal than Mastoid recordings (ICC = 0.86 [95% CI: 0.77–0.90] vs. 0.77 [0.68–0.84]; CV 12–14%). Latency showed excellent agreement (CV 2–3%). Adjusting amplitude for head size and residual noise reduced the central 95% range by 18%, although the distribution remained wide. Tester-related variance remained below residual within-ear variance across all conditions.

Conclusions. Wave I measures show good within-individual reproducibility, supporting longitudinal monitoring. Tester variance was not the dominant source of variability, and the empirically derived SNR threshold and normalization approach provide actionable tools for future multi-tester studies.


Inclusion Visit
(Visit 1)
-Inclusion form
-Audiometric tests
-Questionnaires
Electrophysiological
(Visit 2)
- Skin preparation
- Electrophysiology
- Questionnaires
Tester 1
Electrophysiological
(Visit 3)
- Skin preparation
- Electrophysiology
- Questionnaires
Tester 1
Electrophysiological
(Visit 4)
- Skin preparation
- Electrophysiology
- Questionnaires
Tester 1
Electrophysiological
(Visit 5)
- Skin preparation
- Electrophysiology
- Questionnaires
Tester 2,3,4 or 5


Ear-canal ABR (●)
Mastoid ABR (■)
Good reliability
Excellent reliability
W1 Amplitude
Peak to Trough
0.86
0.77
W1 Amplitude
Trough to Peak
0.82
0.60
W1 Amplitude
Baseline Normalized
0.83
0.68
W1 Latency
0.86
0.70
0.4
0.5
0.6
0.7
0.8
0.9
1.0
ICC(1,1) with 95% Confidence Interval


ICC(2,1)
1.0
0.9
0.8
0.7
0.6
0.5
Excellent reliability
Good reliability
0.80
0.77
0.66
0.74
Ear-canal W1 Latency
Ear-canal W1 Amplitude
Mastoid W1 Latency
Mastoid W1 Amplitude
Ear-canal ABR
Mastoid ABR
Variance explained (%)
100
80
60
40
20
0
11.8%
8.1%
80.1%
13.4%
10.1%
76.5%
26.3%
7.3%
66.4%
23.3%
2.5%
74.2%
Ear-canal W1 Latency
Ear-canal W1 Amplitude
Mastoid W1 Latency
Mastoid W1 Amplitude
Between participants
Operator
Residual

Session 4.A. Machine Learning & AI for Hearing Science

Intelligent algorithms for diagnosis, prediction, and personalisation of hearing interventions.

Chaired by Prof. Dr. Tobias Goehring.

Perceptual Sensitivity is Explained by Optimization for Ecological Hearing Tasks

Mark Saddler (DTU Department of Health Technology)*; Torsten Dau (DTU Department of Health Technology); Josh McDermott (MIT Department of Brain and Cognitive Sciences)

marksa@dtu.dk

Featured talk

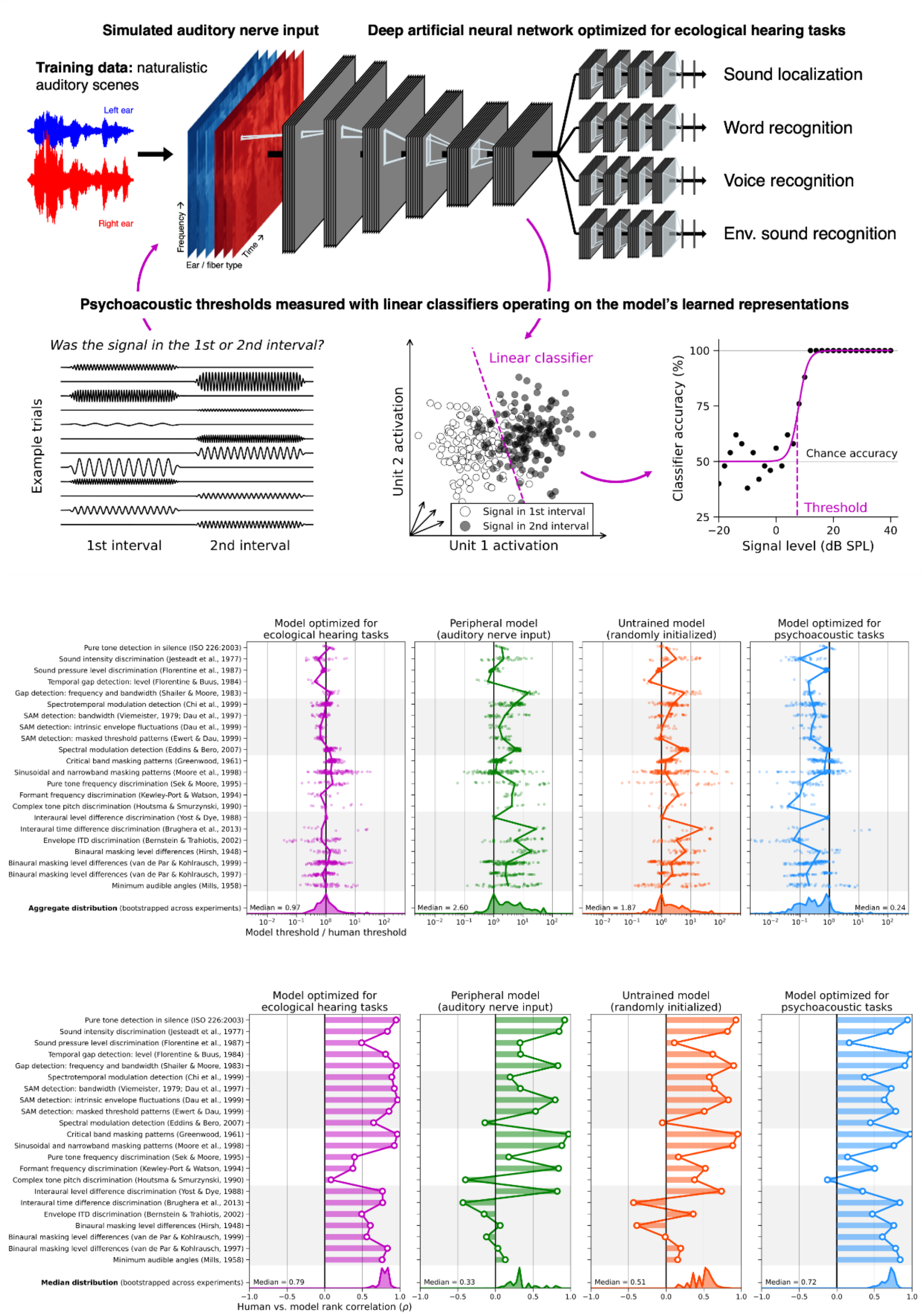
Abstract

Background. When asked to discriminate between two stimuli that differ along some dimension, such as intensity, frequency, or orientation, humans exhibit a smallest stimulus difference, or threshold, below which discrimination becomes impossible. These limits on perceptual sensitivity have traditionally been explained by internal noise. Contemporary theories instead link perceptual sensitivity to efficient coding of the natural environment but have not been comprehensively tested. Here, we measured the psychoacoustic sensitivity of a computational model optimized for everyday hearing tasks.

Methods. We first optimized a deep artificial neural network to jointly localize and recognize words, voices, and environmental sounds using simulated auditory nerve representations as input. We then estimated the model's psychoacoustic thresholds by training linear classifiers to make simple discrimination judgments using the network's learned representations.

Results. Thresholds derived from ecologically task-optimized representations exhibited qualitative and quantitative matches to human sensitivity across a large set of perceptual dimensions. By contrast, thresholds derived from alternate model representations did not match those of humans. Untrained and peripheral representations tended to underestimate human sensitivity, whereas representations specifically optimized for psychoacoustic tasks tended to produce superhuman sensitivity.

Conclusions. The results show that many classical perceptual phenomena (such as spectral/temporal modulation sensitivity and binaural unmasking) are side effects of representations optimized for natural listening behavior, and raise the possibility that psychophysical thresholds are determined by linear separability in these optimized representations, rather than intrinsic neural noise.


Simulated auditory nerve input
Deep artificial neural network optimized for ecological hearing tasks
Training data: naturalistic auditory scenes
Left ear
Right ear
Frequency
Time
Ear / fiber type
Sound localization
Word recognition
Voice recognition
Env. sound recognition
Psychoacoustic thresholds measured with linear classifiers operating on the model's learned representations
Was the signal in the 1st or 2nd interval?
Example trials
1st interval
2nd interval
Linear classifier
Unit 2 activation
Unit 1 activation
Signal in 1st interval
Signal in 2nd interval
Classifier accuracy (%)
Chance accuracy
Threshold
Signal level (dB SPL)
Model optimized for ecological hearing tasks
Peripheral model (auditory nerve input)
Untrained model (randomly initialized)
Model optimized for psychoacoustic tasks
Pure tone detection in silence (ISO 226:2003)
Sound intensity discrimination (Jesteadt et al., 1977)
Sound pressure level discrimination (Florentine et al., 1987)
Temporal gap detection: level (Florentine & Buus, 1984)
Gap detection: frequency and bandwidth (Shailer & Moore, 1983)
Spectrotemporal modulation detection (Chi et al., 1999)
SAM detection: bandwidth (Viemeister, 1979; Dau et al., 1997)
SAM detection: intrinsic envelope fluctuations (Dau et al., 1999)
SAM detection: masked threshold patterns (Ewert & Dau, 1999)
Spectral modulation detection (Eddins & Bero, 2007)
Critical band masking patterns (Greenwood, 1961)
Sinusoidal and narrowband masking patterns (Moore et al., 1998)
Pure tone frequency discrimination (Sek & Moore, 1995)
Formant frequency discrimination (Kewley-Port & Watson, 1994)
Complex tone pitch discrimination (Houtsma & Smurzynski, 1990)
Interaural level difference discrimination (Yost & Dye, 1988)
Interaural time difference discrimination (Brughera et al., 2013)
Envelope ITD discrimination (Bernstein & Trahiotis, 2002)
Binaural masking level differences (Hirsh, 1948)
Binaural masking level differences (van de Par & Kohlrausch, 1999)
Binaural masking level differences (van de Par & Kohlrausch, 1997)
Minimum audible angles (Mills, 1958)
Aggregate distribution (bootstrapped across experiments)
Median = 0.97
Median = 2.60
Median = 1.87
Median = 0.24
Model threshold / human threshold
Median distribution (bootstrapped across experiments)
Median = 0.79
Median = 0.33
Median = 0.51
Median = 0.72
Human vs. model rank correlation (ρ)

Frequency Following Response in Children with School Difficulties or Stroke: Integrating Traditional Statistics and Machine Learning Techniques

Jheniffer Raimundo (UNICAMP)*; Thalita Ubiali (UNICAMP); Betânia Eugenia Rodrigues da Silva (UNICAMP); Rafael Ribeiro Santos (UNICAMP); Edson Borin (UNICAMP); Maria Francisca Colella-Santos (UNICAMP)

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Podium

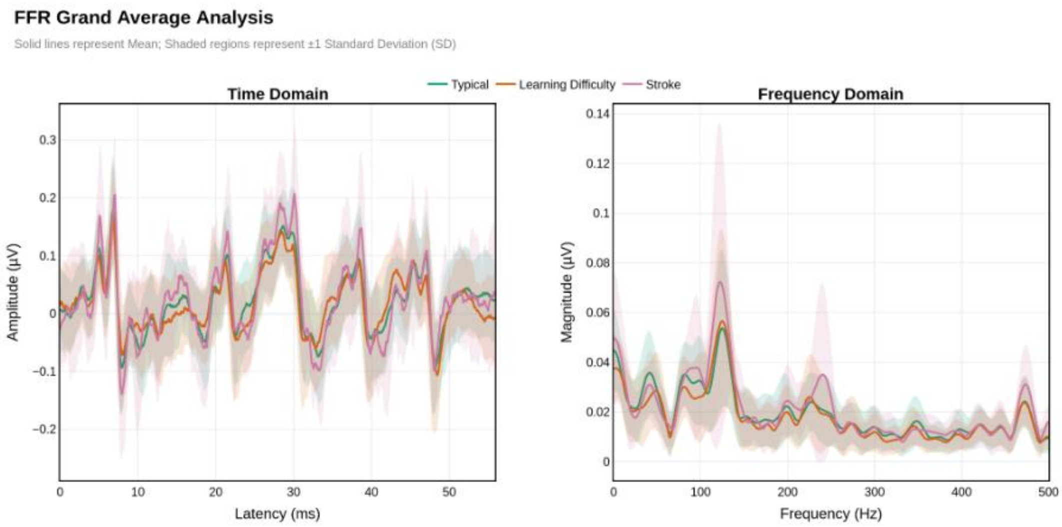
Abstract

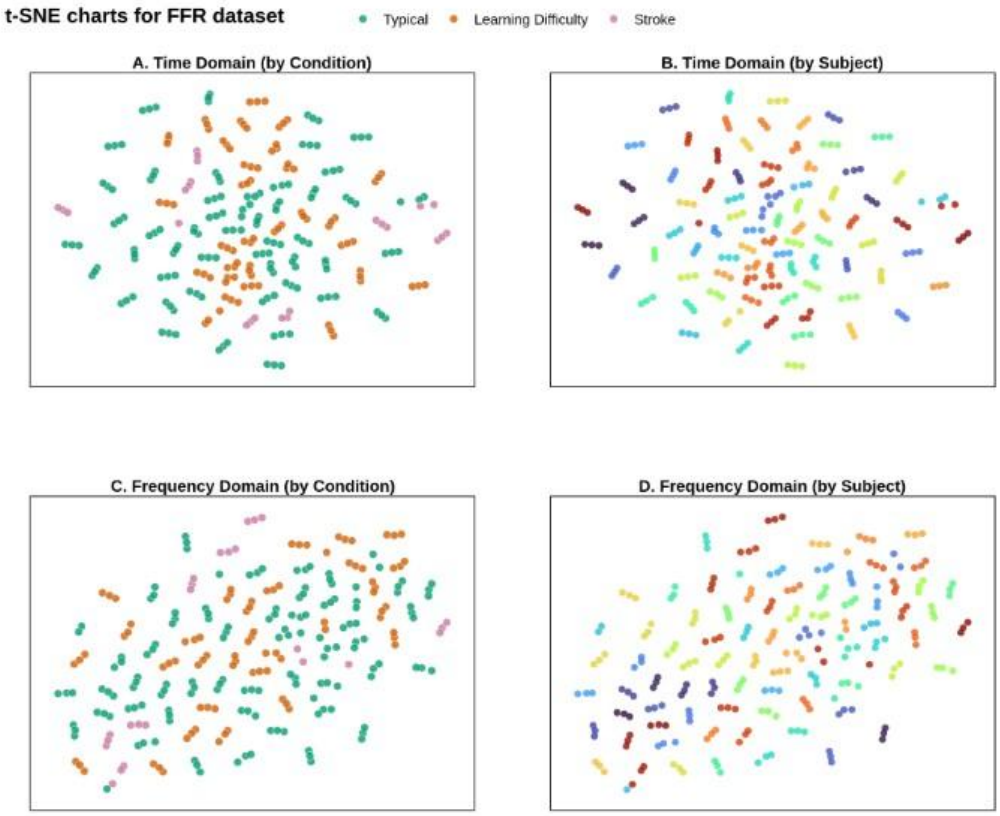
Background. The Frequency Following Response (FFR) provides crucial insights into subcortical auditory processing and fine temporal coding. While traditional statistics are standard for clinical analysis, Artificial Intelligence (AI) offers new possibilities for pattern recognition. This study aims to integrate traditional and computational FFR analyses to provide a comprehensive diagnostic framework for children with school difficulties or stroke sequelae.

Methods. Evaluated 101 children (7–15 years old) divided into three groups: School Difficulties (GE-I, n=33), Stroke (GE-II, n=10), and Typical Development (GC, n=58). FFR was evoked using the /da/ syllable. Statistical analysis employed the Bootstrap technique (95% CI, $p<0.05$). For the computational approach, a pipeline was developed in Python (PyTorch) using supervised and self-supervised machine learning (ML) models, with data partitioning by subject to prevent data leakage.

Results. Time-domain analysis revealed significantly higher amplitudes for peaks D ($p=0.01$) and E ($p=0.02$) in the stroke group (GE-II) compared to others. Frequency-domain analysis also showed higher magnitudes for GE-II. Conversely, machine learning models functioned primarily as "patient detectors" rather than "condition detectors," identifying each subject's unique biological signature. t-SNE visualizations demonstrated strong intra-class consistency (clustering by individual) but a lack of group-based clustering, suggesting that sound encoding is highly individualized and influenced by phonemic and linguistic processing.

Conclusions. While descriptive statistics effectively differentiated the groups by magnitude, ML revealed high signal stability and intra-class consistency. Despite sample size limitations for automated group categorization, AI exploration confirmed that FFR captures a reliable and persistent neural signature. This integrated approach highlights the potential of FFR as a tool for precision and longitudinal monitoring.


FFR Grand Average Analysis
Solid lines represent Mean; Shaded regions represent ±1 Standard Deviation (SD)
Typical
Learning Difficulty
Stroke
Time Domain
Amplitude (µV)
Latency (ms)
Frequency Domain
Magnitude (µV)
Frequency (Hz)


t-SNE charts for FFR dataset
Typical
Learning Difficulty
Stroke
A. Time Domain (by Condition)
B. Time Domain (by Subject)
C. Frequency Domain (by Condition)
D. Frequency Domain (by Subject)

Mapping the Hearing Loss Landscape: A Data-Driven Phenotyping Framework Beyond Pure-Tone Audiometry

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Podium

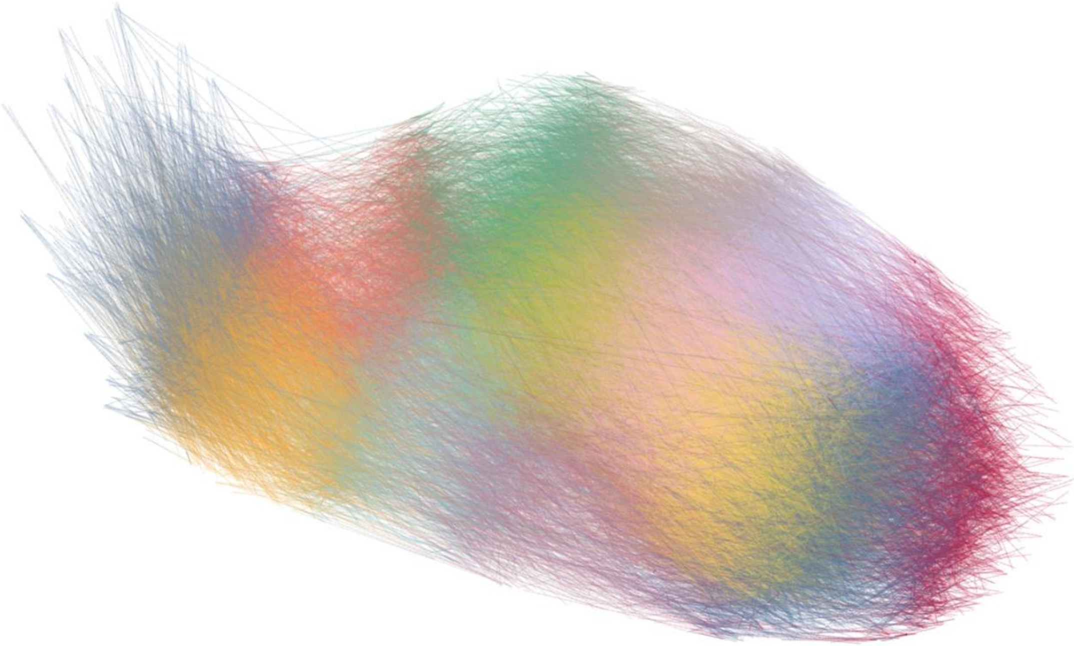
Abstract

Background. Hearing impairment is multifactorial, yet pure-tone audiometry remains the clinical gold standard despite its limited sensitivity to certain pathologies, including hidden hearing damage. Prior phenotyping approaches either relied solely on audiogram data or incorporated additional tests but were constrained by small sample sizes, leaving a substantial portion of the patient space unclassified.

Methods. We analysed a large clinical dataset from Rigshospitalet (Denmark) spanning 1995–2022, comprising ~300,000 audiograms from over ~85,000 unique adult patients, including audiometric thresholds and speech audiometry results. We compared Principal Component Analysis (PCA) and Uniform Manifold Approximation and Projection (UMAP) for two-dimensional visualisation. Clustering algorithms — Gaussian Mixture Models, and Leiden community detection — were applied to a sensorineural hearing loss (SNHL) subset. Longitudinal trajectory analysis was also performed to characterise phenotypic progression over time.

Results. PCA captured major audiogram variance but proved limited for multimodal audiological data. UMAP better exploited both global and local data structures and was adopted as the basis for our Hearing Loss Map. Comparison with existing phenotype classifications revealed that up to 40% of data remained unclassified. Clustering analyses identified novel SNHL phenotypes not captured by conventional schemes, and trajectory analysis enabled longitudinal characterisation of hearing loss progression.

Conclusions. We propose this UMAP-based Hearing Loss Map as a navigational tool for the SNHL patient space, with potential applications in clinical decision-making, patient counselling, evaluation of novel treatments, and development of next-generation hearing technologies.



DeepFS4: A Deep Neural Network-Based Sound Coding Strategy for Cochlear Implants

Xavier Pérez Ruiz (Hannover Medical School)*; Robert Hart (Hannover Medical School); Waldo Nogueira (Hannover Medical School)

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Podium

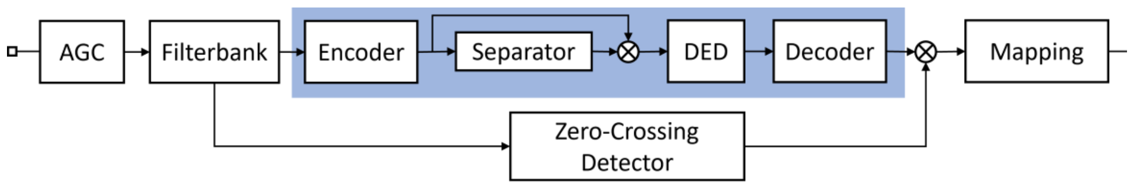
Abstract

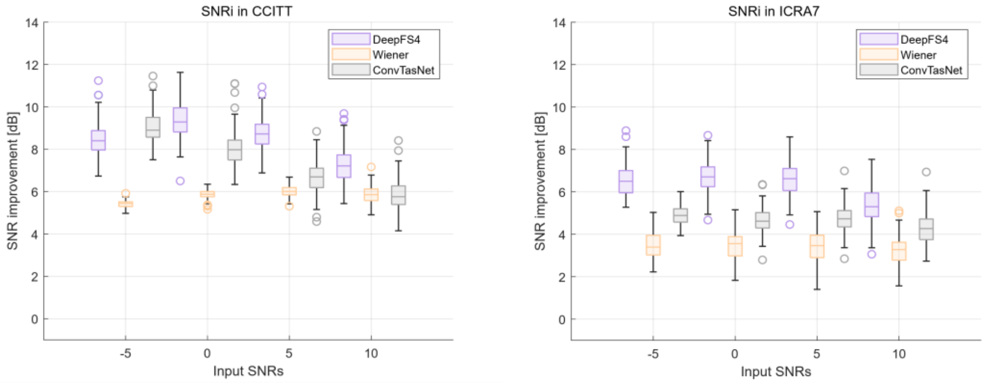
Background. Cochlear implants (CIs) can restore sound perception, yet users struggle with speech understanding in noise. Existing speech enhancement methods often introduce latency and ignore CI coding characteristics. While end-to-end networks like DeepACE improve intelligibility without added latency, they focus on temporal envelopes. Little research has applied this approach to strategies utilizing temporal fine structure (FS), such as FS4, which mimics neural phase locking.

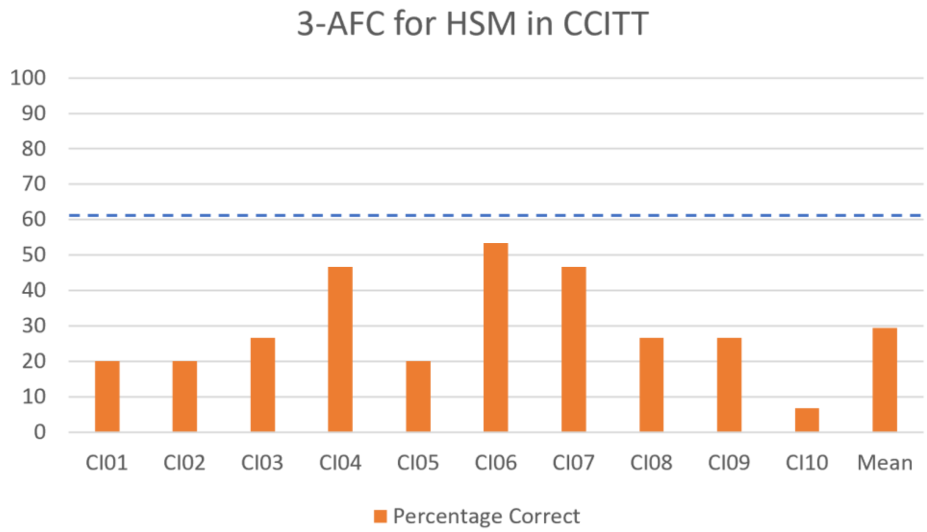
Methods. We developed DeepFS4, an end-to-end version of FS4. We began with a DeepACE-like Temporal Convolutional Network (TCN) that performs envelope-based speech enhancement (SE). DeepFS4 was evaluated using objective measures and speech understanding tests in 10 CI recipients. These included the German HSM sentence test, a MUSHRA quality rating, and a 3-AFC evaluation to examine the effect of FS locations. To create the end-to-end version of FS4, an automatic gain control (AGC) based on a simpler TCN and several convolutional layers representing the finite impulse response (FIR) filterbank were integrated before the main TCN. An additional layer was added in parallel to emulate the zero-crossing detector for FS information.

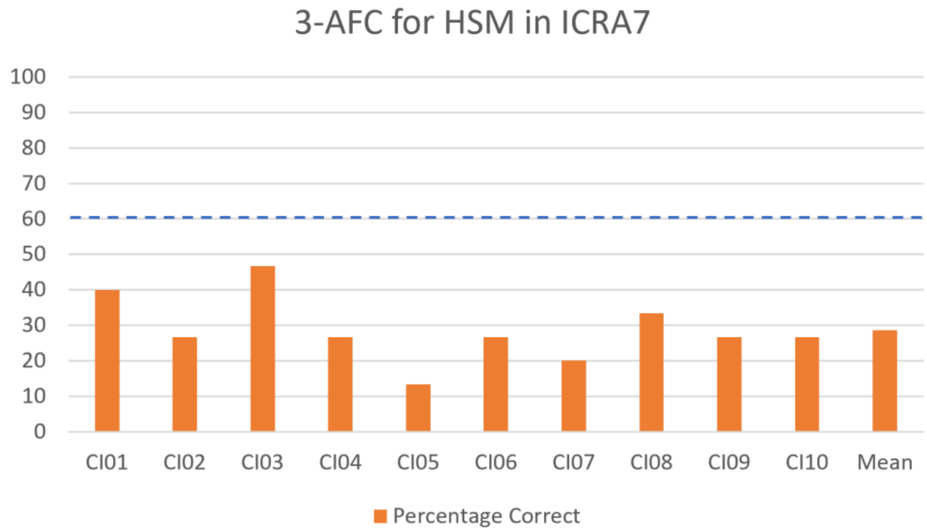
Results. DeepFS4 achieved a mean signal-to-noise ratio improvement (SNRi) of 7 dB. In listening tests, it yielded significantly higher scores than both commercial FS4 and state-of-the-art front-end algorithms. Negligible differences in speech intelligibility were revealed when comparing the use of different FS location variants, i.e., of correctly placed zero-crossings against misplaced (noisy) ones.

Conclusions. DeepFS4 demonstrates that integrating SE directly into sound coding via low-latency deep learning improves CI speech perception in noise. Robust improvements are possible even without FS location enhancements, validating our end-to-end architecture where SE is performed only on the envelope and bringing us closer to AI-powered, clinically viable sound coding.




SNRi in CCITT
SNRi in ICRA7
DeepFS4
Wiener
ConvTasNet
SNR improvement [dB]
Input SNRs


3-AFC for HSM in CCITT
CI01 CI02 CI03 CI04 CI05 CI06 CI07 CI08 CI09 CI10 Mean
Percentage Correct


3-AFC for HSM in ICRA7
CI01 CI02 CI03 CI04 CI05 CI06 CI07 CI08 CI09 CI10 Mean
Percentage Correct

An AI Voice Agent for Speech Communication Assessment

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Podium

Abstract

Background. Although hearing aids (HAs) aim to improve communication, their benefit is usually assessed with non-interactive listening tests. These tests measure hearing thresholds and speech intelligibility, but not communication ability. The need for an assistance-free, scalable, and reproducible speech communication test has been recognized in the literature, but practical challenges have prevented its realization.

Methods. Advances in speech technologies enable the use of AI voice agents as partners in communication testing. We present a prototype of such a system: a web-based voice agent built using a native streaming LLM API that allows bidirectional, low-latency audio interaction. The acoustic scene is managed via the Web Audio API, allowing difficulty to be scaled by adjusting SNR, noise type, and target spatialization. We designed a café role-play task in which participants interact with the agent and collect requested information. Agent behavior and task goals are controlled through prompt engineering. Following the task, participants completed a questionnaire assessing the tool's suitability for future studies. Two normal-hearing groups were tested at SNRs of +5 dB and −5 dB.

Results. Participants rated the interaction as sufficiently natural. At both SNRs, speech latency was judged acceptable and conversational flow was maintained. Perceived difficulty differed between groups, but most participants successfully collected the target details. While this confirms effective communication with the agent, it indicates limited sensitivity of the paradigm to quantify communication difficulties.

Conclusions. Our prototype shows the potential of an AI voice agent for speech communication research. In the future, we plan to use HA and hearing loss simulations and develop a more sensitive paradigm combining an information-gap task with a secondary task. Planned metrics include task scores, clarification requests, repair sentences, and turn-taking analysis.


prompt engineering
text
LLM CHAT
text
voice type, lombard speech emotion, speaking rate
TEXT-TO-SPEECH
SPEECH RECOGNITION
background noise type
acoustic parameters
basic visual components
hello, can you hear me?
not really, the background is too loud...

Inferring Cochlear Cell Damage From Auditory Evoked Potentials Using Physics-Informed Machine Learning.

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Podium

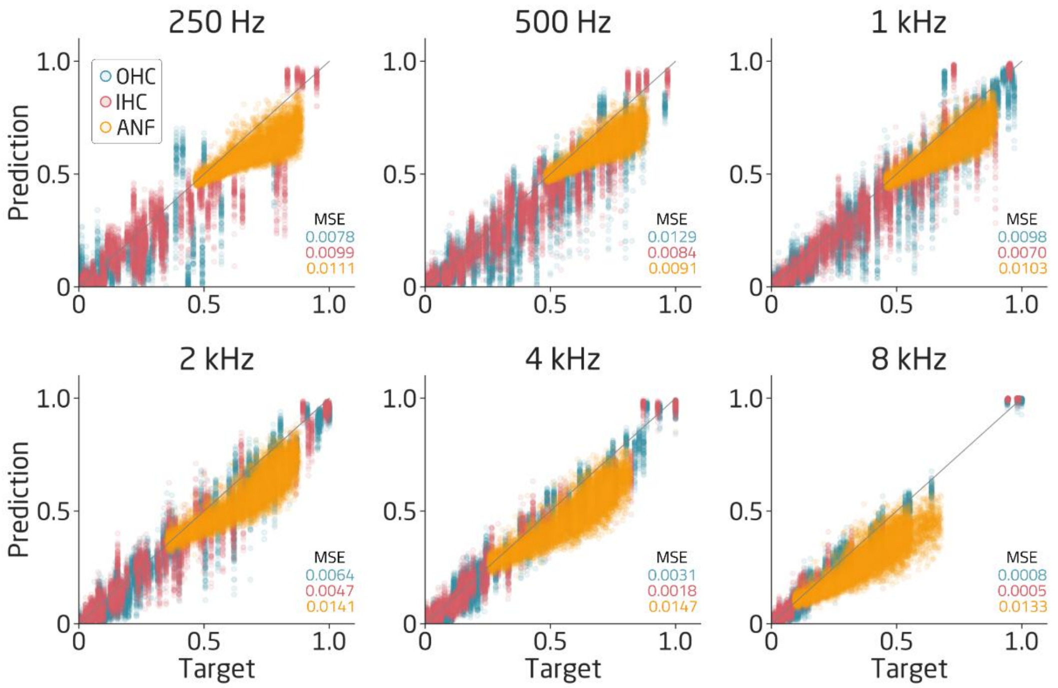
Abstract

Background. Auditory evoked potentials (AEP) are widely used clinically but they offer limited insight into the specific cochlear mechanisms underlying hearing loss. Because outer hair cells (OHC), inner hair cells (IHC), and auditory nerve fibers (ANF) are functionally coupled, it is difficult to disentangle their individual contributions from recorded responses. Computational forward models can link cochlear pathologies to predicted AEPs. Inverting this mapping using machine learning (ML) could enable more precise and individualized cochlear diagnostics.

Methods. A physiologically detailed auditory nerve model was used to simulate AEPs across diverse cochlear damage profiles derived from standard audiograms and human histopathological data. Each profile was encoded as an 18-dimensional vector representing OHC, IHC, and ANF integrity across six cochlear segments. A total of 55,000 profiles, with responses to acoustic stimuli at multiple intensities, were used to train and test a convolutional neural network. The network predicted continuous cell-status values (0-1) for each cochlear cell population and segment.

Results. The network accurately decoded OHC, IHC, and ANF damage from simulated AEPs in ears with previously unseen audiograms (Fig. 1). It could also distinguish different ratios of IHC-to-OHC damage, as well as distinct ANF profiles, even across ears with identical audiograms. Ongoing work identifies a minimal stimulus subset that preserves decoding performance while reducing test time.

Conclusions. We present a physics-informed ML framework for mapping AEPs onto cochlear pathology. Physiologically validated simulations provide labeled training data unavailable from clinical recordings, enabling ML models to identify pathology-specific response signatures. These results demonstrate the feasibility of distinguishing OHC, IHC, and ANF dysfunction non-invasively and represent a step toward precision audiology and pathology-targeted rehabilitation.



Confusion scatter matrix across cochlear frequency regions (250 Hz to 8 kHz). Each panel compares target vs. predicted cell-status values for OHC (blue), IHC (red), and ANF (orange), with the gray diagonal indicating perfect agreement; denser points near the diagonal indicate lower prediction error. Mean square error (MSE) indicated in each panel.

Predicting Early Cochlear Implant Outcomes in Adults: A Machine Learning Approach Using Cognitive and Linguistic Measures

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Poster

Abstract

Background. Speech recognition (SR) outcomes after cochlear implantation vary widely, motivating efforts to identify pre-operative (pre-op) variables predicting post-cochlear implant (CI) performance. Cognitive and linguistic (CL) abilities show promise as such predictors but remain largely unexplored. To determine if CL abilities improve post-CI SR prediction over conventional pre-op variables alone, we compared machine learning models with and without CL variables.

Method: Pre-op and 1-month post-CI SR measures (CNC words, AzBio sentences in quiet and noise) were collected from 42 adult postlingually deaf CI candidates with bilateral moderate-to-profound sensorineural hearing loss, along with demographic and pre-op CL variables. CL variables included Mini Mental State Examination (MMSE), Wide Range Achievement Test (WRAT) word reading, lexical and phonological access speed on Test of Word Reading Efficiency (TOWRE), nonverbal reasoning in Raven's progressive matrices, and inhibition-concentration via visual Stroop color-word test. LASSO-regularized logistic regression models were fit with and without CL variables to classify post-CI SR into high, medium, and low performance groups.

Results. Most models incorporating CL variables demonstrated significantly improved predictions, with 21%, 72%, and 4% increases in macro-averaged AUROC for CNC bilateral best-aided, CNC CI-only, and AzBio in noise, respectively. The AzBio in quiet model did not benefit from including CL variables. CL feature selection frequency, as determined by LASSO, varied by outcome measure, with CNC models selecting CL variables 88% more frequently than AzBio models.

Conclusion: These findings suggest pre-op CL abilities contribute meaningful predictive value for early post-CI SR beyond conventional variables, particularly for word recognition. Considering CL abilities in pre-op CI candidacy evaluations may improve patient stratification and help set individualized expectations for post-CI SR.

Post-CI Outcome Measure	CNC Best-aided	CNC CI-only	AzBio In Quiet	AzBio In Noise
macroAUC (w/ CL variables)	**0.602** ±0.032	**0.627** ±0.073	0.579 ±0.028	**0.741** ±0.034
macroAUC (w/o CL variables)	0.498 ±0.027	0.365 ±0.074	0.622 ±0.034	0.714 ±0.029

Table 1: Comparison of macro-averaged AUROC for each post-CI speech recognition outcome model with and without cognitive and linguistic (CL) variables. Results are averaged over multiple model fitting runs with different random seed initializations, and standard deviations are listed below each average. Bolded values are statistically significant in paired t-tests.

	Age	Hearing Loss Duration	Sex	Implant Side	Implant Brand	Socioeconomic Score	Better Ear Pure Tone Avg.	Pre-CI AzBio in Quiet	WRAT	Raven's Prog. Matrices	Stroop Congruent	Stroop Incongruent	Stroop Control	TOWRE Sight Word Eff.	TOWRE Phonemic Decoding Eff.	MMSE
CNC Bilateral Best-Aided (w/ CL)	0.13	0.35	0.02	0.34	0.08	0.36	0.40	0.41	0.03	0.63	0.55	0.38	0.35	0.36	0.17	0.67
CNC Bitateral Best-Aided (w/o CL)	0.10	0.09	0.02	0.37	0.34	0.47	0.62	0.67								
CNC CI-Only (w/ CL)	0.17	0.04	0.04	0.25	0.22	0.29	0.42	0.36	0.07	0.33	0.23	0.17	0.25	0.35	0.58	0.33
CNC CI-Only (w/o CL)	0.26	0.22	0.22	0.29	0.25	0.33	0.42	0.39								
AzBio in Quiet (w/ CL)	0.25	0.15	0.00	0.01	0.09	0.35	0.26	0.67	0.02	0.46	0.26	0.20	0.03	0.11	0.02	0.60
AzBio in Quiet (w/o CL)	0.32	0.23	0.00	0.04	0.38	0.46	0.42	0.67								
AzBio in Noise (w/ CL)	0.24	0.22	0.00	0.00	0.00	0.00	0.35	0.67	0.23	0.19	0.00	0.14	0.00	0.14	0.06	0.44
AzBio in Noise (w/o CL)	0.27	0.34	0.00	0.00	0.19	0.05	0.50	0.67								

Figure 1: Frequencies of feature selections during model fitting with LASSO regularization for each post-CI speech recognition outcome measure, with and without cognitive and linguistic (CL) variables. Lower frequencies are shaded in blue hues and higher frequencies are shaded in orange hues.

Machine Learning-Based Prediction of High-Frequency Hearing Risk in Airport Catering Personnel Using Demographic, Occupational, and Baseline Audiometric Variables

Şeyma Öztürk (Istanbul Medipol University)*; Halil Berkay Saldırım (Istanbul Medipol University)

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Poster

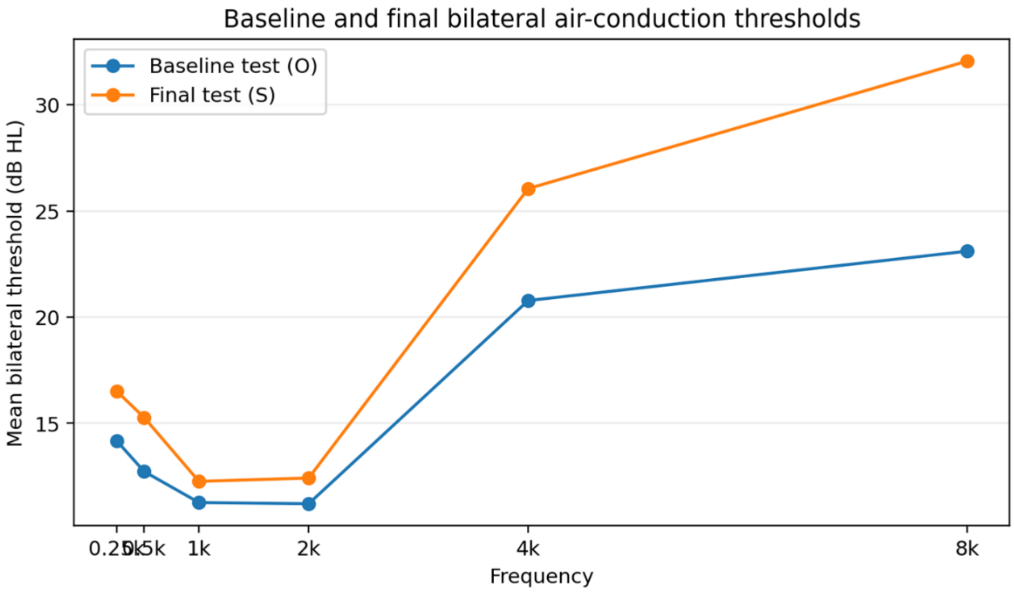
Abstract

Background. Occupational audiometry detects high-frequency threshold shifts but rarely supports individual risk stratification before deterioration is established. We tested whether demographic, occupational and baseline audiometric data predict final high-frequency hearing risk in airport catering personnel.

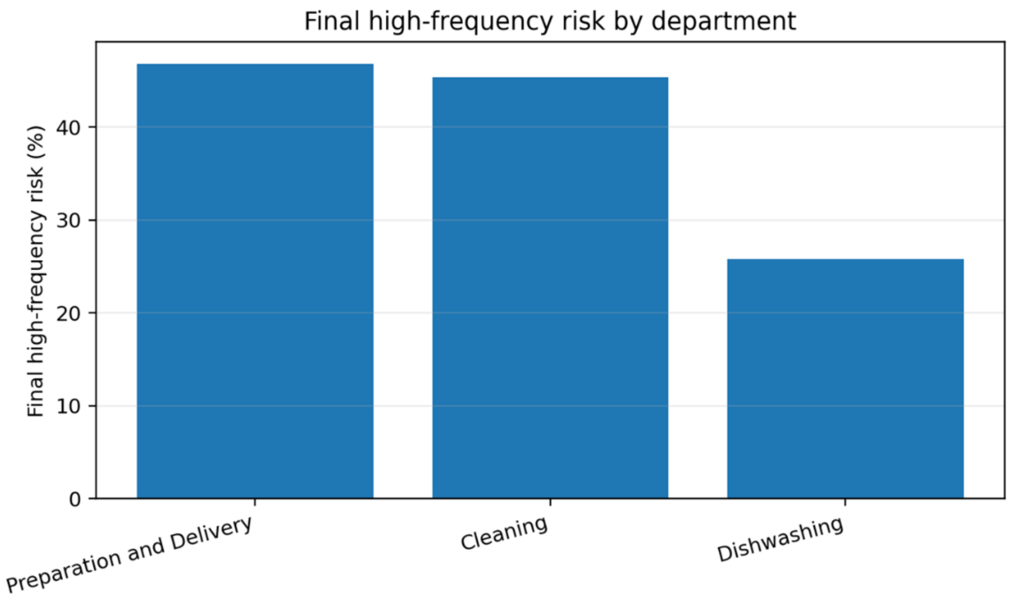
Methods. This retrospective analysis included 512 workers with baseline and final air-conduction thresholds at 0.25–8 kHz in both ears. Final high-frequency risk was defined as bilateral 4–8 kHz PTA ≥30 dB HL. Logistic regression, random forest and gradient boosting classifiers were evaluated with 5-fold stratified cross-validation. We compared a demographic/occupational model (age, sex, employment duration, department) with an expanded model adding baseline low-frequency PTA, high-frequency PTA, high–low frequency gap and high-frequency interaural asymmetry.

Results. Final high-frequency risk was present in 204/512 workers (39.8%), compared with 106/512 (20.7%) at baseline. Mean high-frequency PTA increased from 21.9 to 29.1 dB HL. Demographic/occupational variables alone showed moderate discrimination (best ROC-AUC 0.724). Adding baseline audiometric features markedly improved prediction: the preferred gradient boosting model achieved ROC-AUC 0.878, balanced accuracy 0.772, specificity 0.893, precision 0.801 and F1 0.719. Baseline high-frequency PTA was the dominant predictor, followed by employment duration. Observed risk increased from 4.7% to 89.8% across the lowest to highest predicted-risk quartiles.

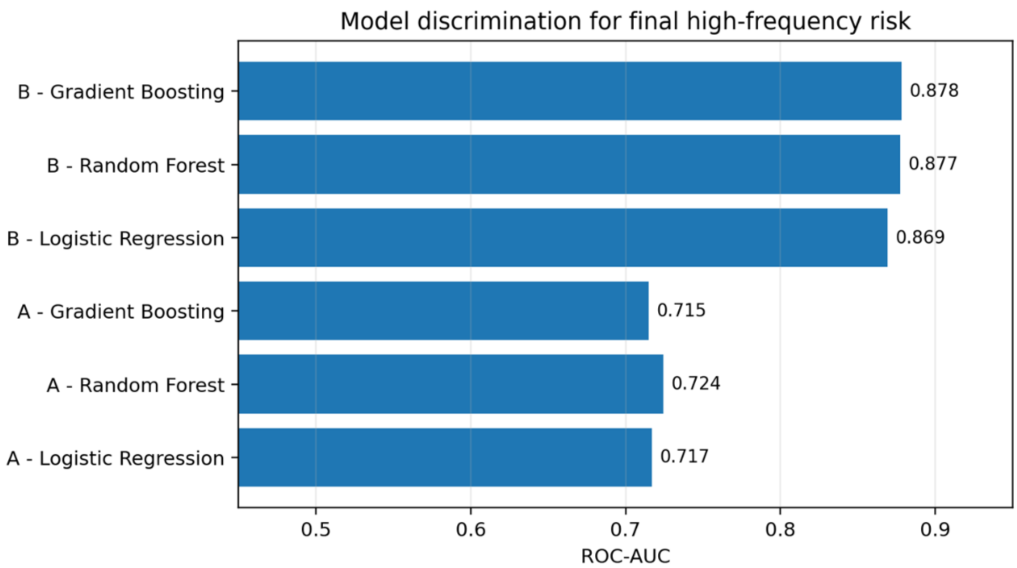
Conclusions. Machine-learning models using routine occupational audiometry can stratify airport catering workers by high-frequency hearing risk. Baseline audiometric profile improves performance beyond demographic and job-related variables, supporting a screening tool for prioritizing follow-up audiometry, counselling and hearing-protection interventions. External validation is required before deployment.



Baseline and final bilateral air-conduction thresholds
Baseline test (O)
Final test (S)
Mean bilateral threshold (dB HL)
30
25
20
15
0.25k
0.5k
1k
2k
4k
8k
Frequency



Final high-frequency risk by department
Final high-frequency risk (%)
40
30
20
10
0
Preparation and Delivery
Cleaning
Dishwashing



Model discrimination for final high-frequency risk
B - Gradient Boosting
0.878
B - Random Forest
0.877
B - Logistic Regression
0.869
A - Gradient Boosting
0.715
A - Random Forest
0.724
A - Logistic Regression
0.717
0.5
0.6
0.7
0.8
0.9
ROC-AUC

EdgeConvGRU: A Lightweight Hybrid Architecture for Robust Acoustic Scene Classification for Medical Hearing Aids

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Poster

Abstract

Background. Hearing aid performance depends strongly on the surrounding acoustic environment, where factors such as noise, reverberation, and competing sound sources directly affect speech intelligibility and listening effort. Traditional acoustic scene classification (ASC) systems rely on computationally intensive deep learning models and semantic environment labels (e.g., “street” or “airport”), which are not well aligned with perceptual needs in hearing aid applications and are difficult to deploy on resource-constrained devices . To address these limitations, we propose a hearing-aid–oriented ASC framework that defines scene categories based on acoustic conditions impacting speech perception, enabling more meaningful and adaptive device behavior.

Method: We introduce EdgeConvGRU, a lightweight hybrid architecture combining convolutional neural networks (CNNs) for spectral feature extraction and gated recurrent units (GRUs) for temporal modeling. Audio signals are converted into log-Mel spectrograms to capture perceptually relevant features, and the model processes short audio segments for real-time inference. The architecture is designed with only 34K parameters to ensure low computational complexity, making it suitable for embedded deployment in hearing aids .

Results. The proposed model performed well on benchmark acoustic scene classification tasks, achieving results comparable to larger state-of-the-art models while maintaining a significantly smaller footprint. Despite having only 34K parameters, EdgeConvGRU demonstrates strong classification performance and remains efficient after quantization, highlighting its suitability for resource-constrained deployment .

Conclusion: EdgeConvGRU demonstrates that efficient, perceptually driven acoustic scene classification is feasible for real-time deployment on resource-constrained hearing aids.

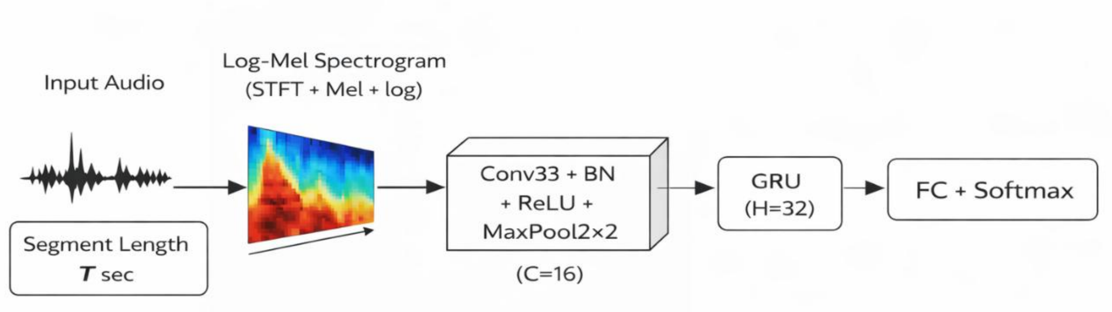

Table 1: *Model Comparison on TUT Acoustic Scene Classification across 3 runs*

Model Type	Accuracy (%)	Params (K)
Gaussian Mixture Model	72.5 ± 0.116	50
MLP classifier	74.8 ± 0.215	300
Multitask CNN ASC [23]	$\mathbf{81.4 \pm 0.3}$	1200
Deep CNN Ensemble [24]	79.0 ± 0.04	2000
EdgeConvGRU (ours)	77.4 ± 0.13	**34**

Intelligent System for the Classification and Segmentation of Middle Ear Pathologies

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Poster

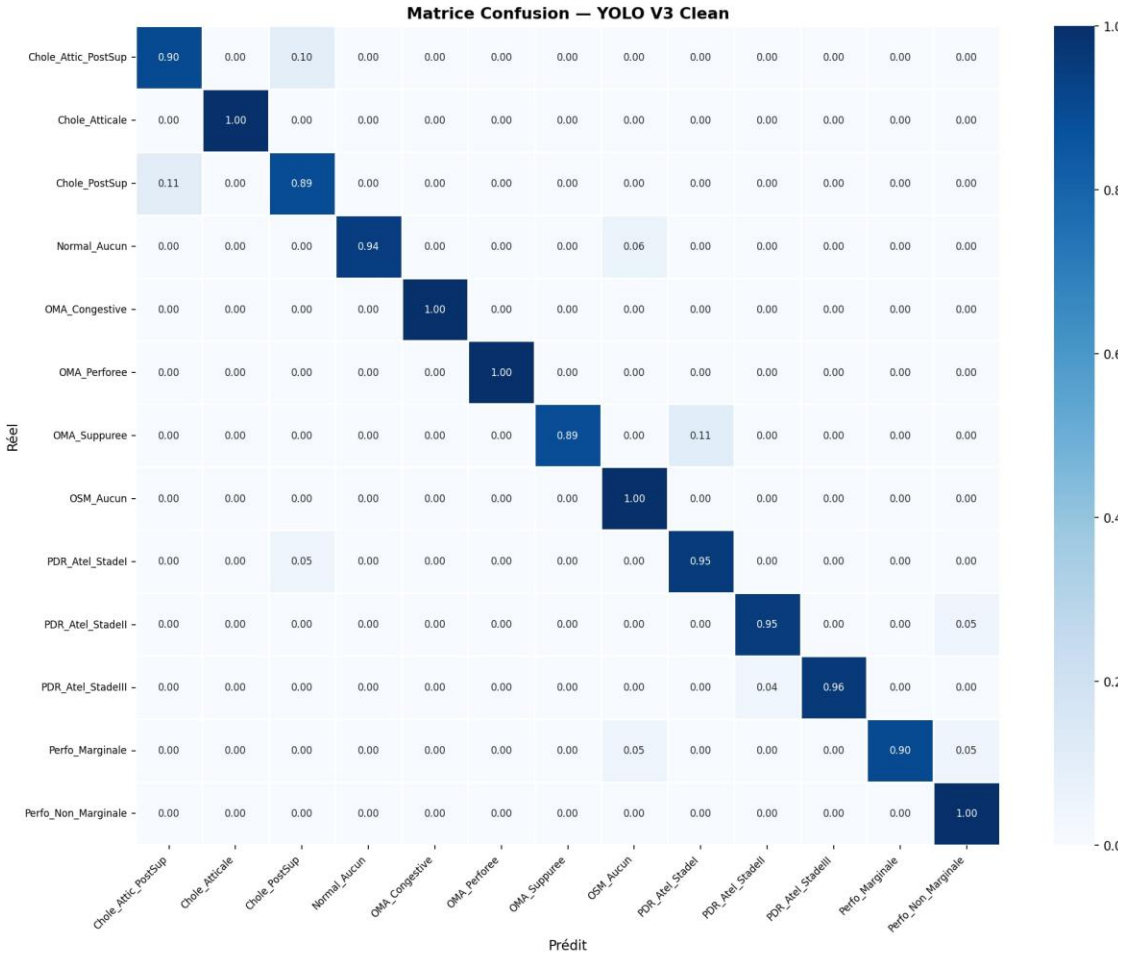
Abstract

Background. The accurate diagnosis of ear pathologies—such as Acute Otitis Media (OMA), Otitis Media with Effusion (OSM), and Cholesteatoma—is critical for effective clinical intervention. In the Tunisian healthcare context, this project represents a significant innovation, being one of the first locally developed AI-driven platforms integrated with clinical data from the Habib Bourguiba University Hospital in Sfax.

Methods. This research develops a dual-stage deep learning framework. For classification, we conducted an extensive benchmark of several architectures, including EfficientNet and various YOLO versions. Unlike standard systems, our classification does not merely identify isolated diseases but also recognizes types and complex combinations, such as Retraction Pockets combined with Cholesteatoma (PDR + Chole). For segmentation, we evaluated multiple models ranging from traditional Convolutional Neural Networks (CNNs) to Transformer-based architectures. The final pipeline utilizes a hybrid approach combining U-Net and SegFormer to leverage both local spatial features and global transformer-based context.

Results. The classification module achieved a high performance with a peak accuracy of 95.8%, outperforming other tested models. The segmentation pipeline, through the fusion of U-Net and SegFormer, reached a mean Intersection over Union (mIoU) of 60%. These results demonstrate a robust reliability in identifying complex pathological co-occurrences and delineating anatomical structures in otoscopic images.

Conclusions. The integration of state-of-the-art architectures like YOLO and SegFormer proves the potential of AI to enhance precision in otology within the Tunisian medical landscape. This platform provides a robust secondary diagnostic opinion, supporting clinical workflows and improving patient outcomes in ear care.


Matrice Confusion — YOLO V3 Clean
Réel
Prédit
Chole_Attic_PostSup
Chole_Atticale
Chole_PostSup
Normal_Aucun
OMA_Congestive
OMA_Perforee
OMA_Suppuree
OSM_Aucun
PDR_Atel_StadeI
PDR_Atel_StadeII
PDR_Atel_StadeIII
Perfo_Marginale
Perfo_Non_Marginale

The Differentiable Auditory Loop (DAL): Neural Activity Alignment for Personalized Hearing Aids

Alejandro Ballesta Rosen (Google Research); Jason Mikiel-Hunter (Macquarie University)*; Julian Maclaren (Google Research); Jack Collins (Google Research); Simon Carlile (Google Research); Richard Lyon (Google Research)

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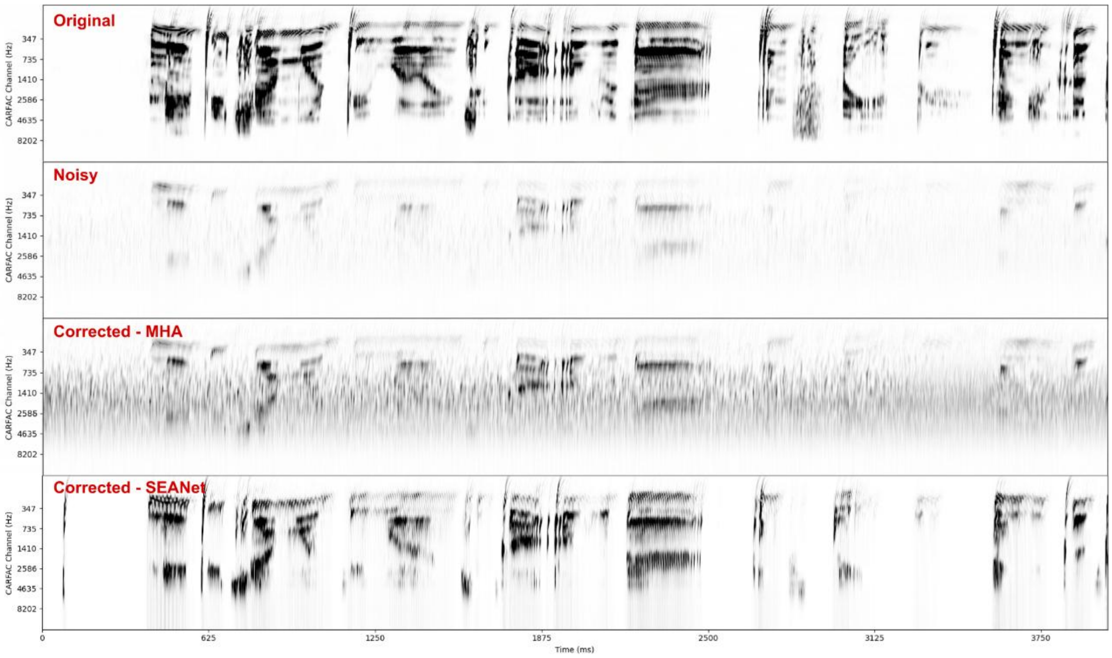
Poster

Abstract

Conventional hearing aids rely on fixed, frequency-dependent amplification and compression to manage reduced sensitivity, which often fails to provide sufficient listening support in complex environments, such as situations with multiple speakers (the “cocktail party” problem). To more comprehensively address the underlying encoding dysfunctions of hearing loss, we introduce the Differentiable Auditory Loop (DAL), a new open-source framework for personalized hearing aid design and fitting. Our first implementation of DAL incorporates CARFAC, a differentiable model of human cochlear function, which we ported to JAX, to optimize a deep neural network to match impaired auditory neural activity patterns with a normal hearing reference. To build a hearing aid with the fine-grained spectro-temporal signal processing required, we adopt SEANet, a waveform-to-waveform fully convolutional UNet generator. We fine-tune the network by comparing the outputs of a CARFAC model fitted to normal hearing with that of a CARFAC model fitted to match each subject’s individual hearing impairment. The comparison is done using a loss function derived from the respective stabilized auditory images (SAIs), which are a 2D representation that captures the fine temporal structure of the auditory nerve output. Through gradient descent, the SEANet model learns to both denoise the input and compensate for the hearing loss modelled by the hearing impaired CARFAC model. Across five metrics measuring both audio interpretability and neural activity recovery, the DAL-optimized ML hearing aid outperformed a conventional master hearing aid (MHA) baseline. The DAL framework provides a novel and practical approach to hearing aid design and fitting that leverages machine learning to enable greater personalization than the state of the art. Next steps include hardware deployment to enable real-world clinical testing.



Visual comparison of CARFAC model output, Neural Activity Patterns (NAPs).

Top to bottom:

(1) Original: clean audio processed through normal-hearing CARFAC (used as reference in DAL loop);

(2) Noisy: noisy audio through an impaired CARFAC (input to DAL loop);

(3) Corrected - MHA: MHA-processed noisy audio (based on NAL-NL2 prescription) through impaired CARFAC

(4) Corrected - SEANet: SEANet-enhanced noisy audio through impaired CARFAC (output of SEANet trained by DAL loop).

CARFAC impairment is applied as OHC health value of 0.5 across all CARFAC channels and is equivalent to mild hearing loss.
The SEANet output uniquely suppresses background noise and closely reconstructs the auditory structural features present in the clean baseline

Session 4.B. Tele-Audiology and Next-Generation Clinical Tools

Remote assessment, AI-assisted monitoring, and digital rehabilitation platforms.

Chaired by Dr. Nikki Philpott.

VIA+: A Gamified Digital Ecosystem for the Integrated Phenotyping of Auditory and Cognitive Development

Frank Betances Reinoso (Sescam HUG)*

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Featured talk

Abstract

Background. Traditional pediatric audiology uses fragmented analogue tests lacking temporal precision and failing to capture hearing-language-cognition interplay. VIA+ (Valoración Interactiva de Audición y Lenguaje) addresses this gap as a modular mHealth platform for integrated multi-domain early childhood assessment.

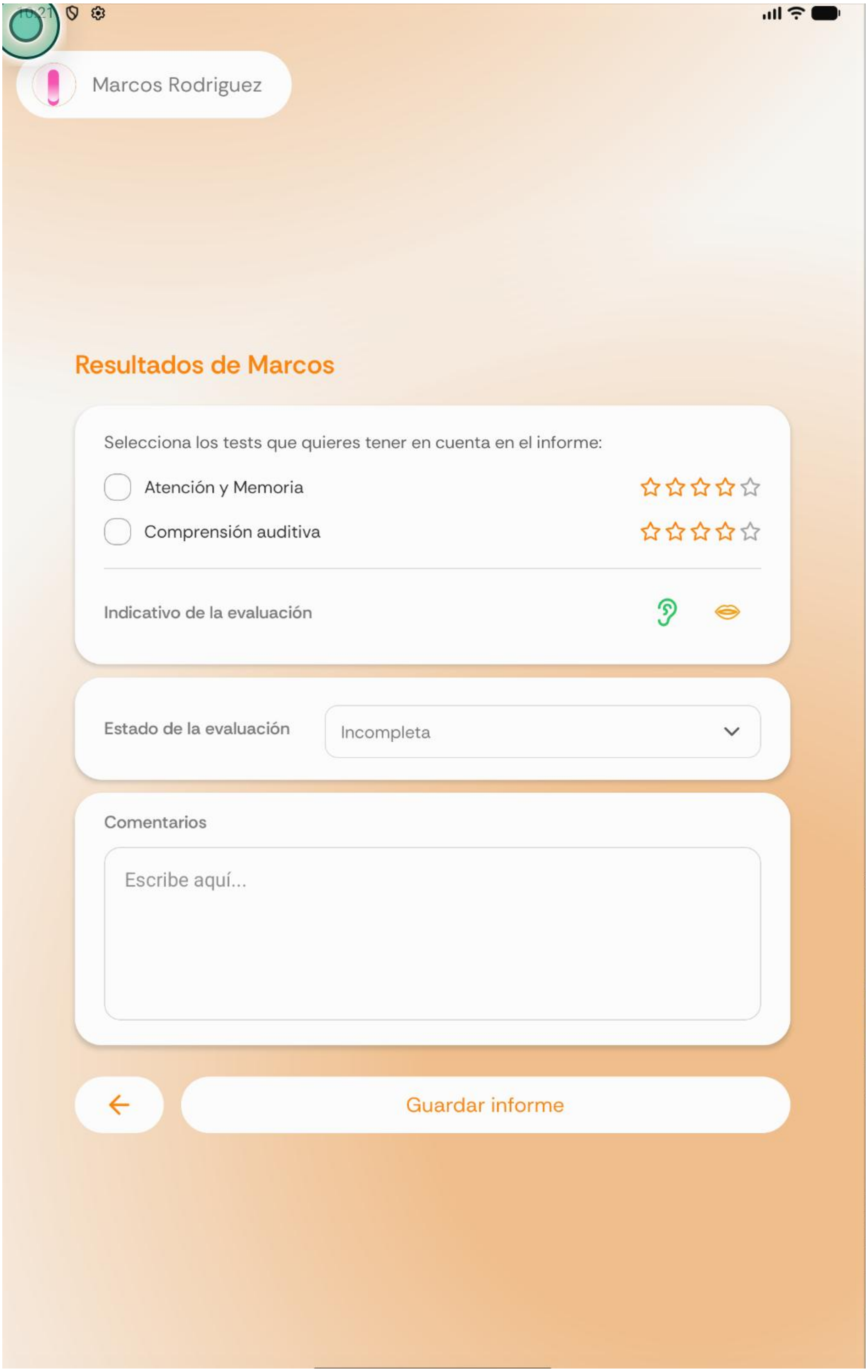
Methods. VIA+ combines an Android tablet app with a secure clinician dashboard, digitizing three domains via seven gamified algorithms: 1) Auditory Processing—automated Conditioned Play Audiometry (CPA) with calibrated stimuli and sound localization; 2) Linguistic Analysis—digital speech audiometry with randomized word-picture matching and audio recording; 3) Cognitive Phenotyping—reaction-time tasks for inhibition (Go/No-Go) and working memory. The system logs response accuracy, latency (ms), and error patterns, storing data locally on the device for subsequent clinical review.

Results. Proof-of-concept validation showed technical stability on Android devices and successful local data storage. The gamified interface sustained attention in 3-7 year-olds, reducing abandonment. Digital latency capture provided sensitive biomarkers distinguishing auditory detection from cognitive delays impossible with manual testing.

Conclusions. VIA+ translates complex protocols into scalable digital tools, unifying auditory-cognitive data to identify neurodevelopmental biomarkers. Local data storage ensures privacy compliance while enabling large-scale, decentralized collection and future AI-driven pediatric hearing analysis.


10:21
Via+
Te damos la bienvenida
A continuación, vamos a preparar la app para tus evaluaciones.
¿Desde qué centro vas a hacer las pruebas?
test
Continuar


10:20
Marcos Rodriguez
Marcos Lopez
Completado
Atención y Memoria
Atención y memoria de trabajo con fichas
Detección de Sonidos
Detección de sonidos
Completado
Comprensión auditiva
Discriminación auditiva de palabras: el niño escucha un sonido y debe identificar la imagen correspondiente.
Repetición Verbal
Imitación verbal o repetición de sílabas/palabras
2 Test realizados
Finalizar evaluación
Comprensión de ordenes y cambios de regla


Marcos Rodriguez
Resultados de Marcos
Selecciona los tests que quieres tener en cuenta en el informe:
Atención y Memoria
Comprensión auditiva
Indicativo de la evaluación
Estado de la evaluación
Incompleta
Comentarios
Escribe aquí...
Guardar informe

A Digital Ecosystem for Scalable Tinnitus Rehabilitation

Zahi Tubul (Kaplan Medical Center)*

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Podium

Abstract

Background. Tinnitus is a prevalent auditory condition that can substantially affect sleep, concentration, emotional well-being, and daily functioning. Evidence-based management approaches—including psychoeducation, cognitive-behavioural strategies, mindfulness-based attention regulation, and sound therapy—have demonstrated clinical value, yet structured tinnitus rehabilitation remains difficult to access in many healthcare systems. Recent developments in digital health technologies create new opportunities to expand integrated tinnitus care delivery.

Methods. This work proposes a conceptual digital framework for tinnitus rehabilitation that delivers established therapeutic domains through a clinician-guided digital platform. Drawing on current tinnitus literature and models of behavioural intervention, the framework integrates four complementary domains: psychoeducation, cognitive-behavioural strategies, mindfulness-based attention regulation, and sound-based approaches. The model emphasises the interaction and synergy between these domains within a structured digital environment.

Results. The proposed ecosystem illustrates how digital platforms can enable scalable, structured tinnitus rehabilitation while preserving clinical guidance. By combining educational content, cognitive strategies, attentional training, and sound-based support within a single digital environment, the framework highlights pathways through which digital tools may facilitate patient engagement and support adaptive coping processes.

Conclusions. Conceptualising tinnitus rehabilitation as a digitally delivered ecosystem of complementary therapeutic domains may help bridge the gap between established clinical knowledge and scalable care delivery. Such frameworks can inform the design of future digital interventions and support the translation of tinnitus research into accessible rehabilitation strategies.


Integrated Digital Tinnitus Rehabilitation
A multi-modal therapeutic ecosystem delivered through a clinician-guided digital platform
Psychoeducation
Understanding tinnitus mechanisms
Sound Therapy
Auditory enrichment and sound-based coping
Clinician-Guided Digital Platform
Cognitive-Behavioral Strategies
Changing maladaptive tinnitus-related thoughts
Mindfulness & Attention Regulation
Reducing attentional capture by tinnitus
Patient Outcomes
Reduced tinnitus salience
Improved coping
Enhanced quality of life

ULIXES: Universal Language-Independent eXploration of Effects of Speech Cues: Pilot Evidence of SNR-Dependent Multisensory Reweighting as a Novel Clinical Index

Jorge Mejia (NAL)*; Leanne Pul (Radboud universitair medisch centrum, Nijmegen); Jan-Willem Wasmann (Radboud universitair medisch centrum, Nijmegen); Noortje van der Mast (Radboud universitair medisch centrum, Nijmegen); Cris Lanting (Radboud universitair medisch centrum, Nijmegen); Marc van Wanrooij (Radboud University, Nijmegen)

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Podium

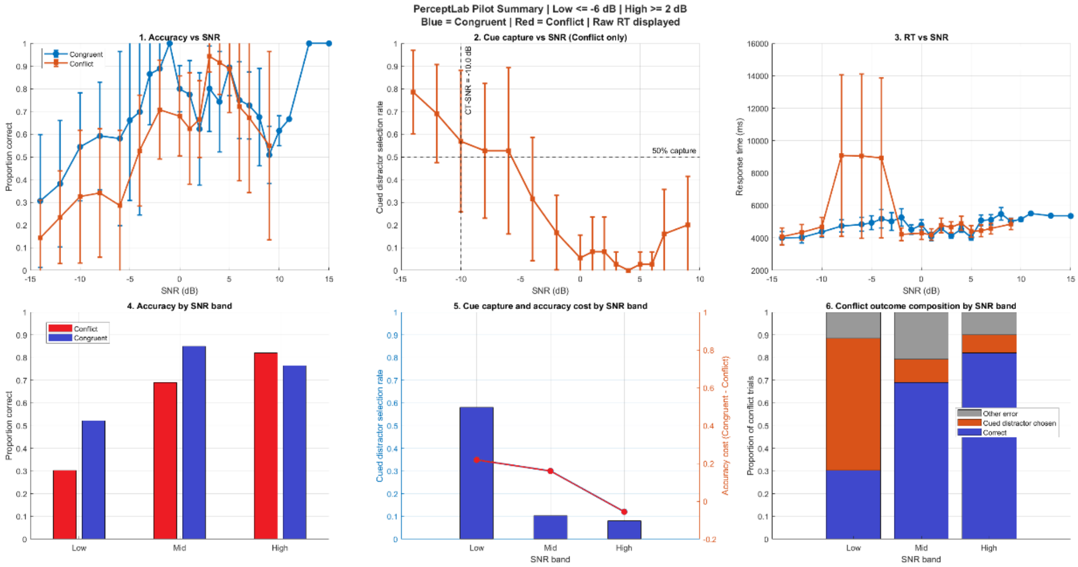
Abstract

Background. Standard speech-in-noise tests do not measure how listeners combine auditory and non-auditory cues in real-world communication. People with hearing loss may rely more on visual or contextual cues, but this cue weighting is not routinely assessed. ULIXES, built on NAL's ULI-test, is a home-based, tablet-compatible speech assessment designed to quantify textual visual capture. Congruent written cues may support perception, while conflicting cues may bias responses toward the distractor. We tested whether capture increases at poorer SNRs and decreases as speech becomes clearer, consistent with reliability-weighted cue use. A capture transition SNR (CT-SNR) was derived as a candidate clinical index.

Methods. In a pilot study (N = 4; 372 trials), participants listened to CVC non-words in noise while viewing written options. The written leading cue either matched the target (congruent; 250 trials) or conflicted with it (conflict; 122 trials). SNR varied adaptively from +12 to -15 dB. Outcomes were accuracy, conflict-trial visual distractor capture, and response time. SNR bands were Low (≤0 dB), Mid (+1 to +4 dB), and High (>+4 dB). Mixed-effects models tested SNR and trial-type effects. CT-SNR was defined as the SNR where capture reached 50%. Data collection is ongoing.

Results. Visual capture showed a clear SNR-dependent pattern. On conflict trials, capture was highest at low SNR (58%) and declined at high SNR (8%; $p \approx 10^{-5}$). Conflict-trial accuracy was lower than congruent-trial accuracy in noise (30% vs 52%) but converged at high SNR (≈82% for both). Models are exploratory.

Conclusions. These pilot data show that textual cues can systematically bias speech reports in noise. Conflict-trial distractor capture was the clearest marker of cue reliance across SNRs. The capture-SNR function and CT-SNR may provide useful measures of multisensory cue weighting beyond accuracy alone. The language-independent CVC non-word format supports broad clinical use.


PerceptLab Pilot Summary | Low <= -6 dB | High >= 2 dB
Blue = Congruent | Red = Conflict | Raw RT displayed
1. Accuracy vs SNR
Congruent
Conflict
Proportion correct
SNR (dB)
2. Cue capture vs SNR (Conflict only)
CT-SNR = -10.0 dB
50% capture
Cued distractor selection rate
SNR (dB)
3. RT vs SNR
Response time (ms)
SNR (dB)
4. Accuracy by SNR band
Conflict
Congruent
Proportion correct
Low
Mid
High
5. Cue capture and accuracy cost by SNR band
Cued distractor selection rate
Accuracy cost (Congruent - Conflict)
Low
Mid
High
SNR band
6. Conflict outcome composition by SNR band
Other error
Cued distractor chosen
Correct
Proportion of conflict trials
Low
Mid
High
SNR band

Development, Implementation, and Clinical Benefits of a Sound Preference Tool

Frederic Marmel (ORCA Labs)*; Dina Lelic (ORCA Labs); Laura Balling (WS Audiology)

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Podium

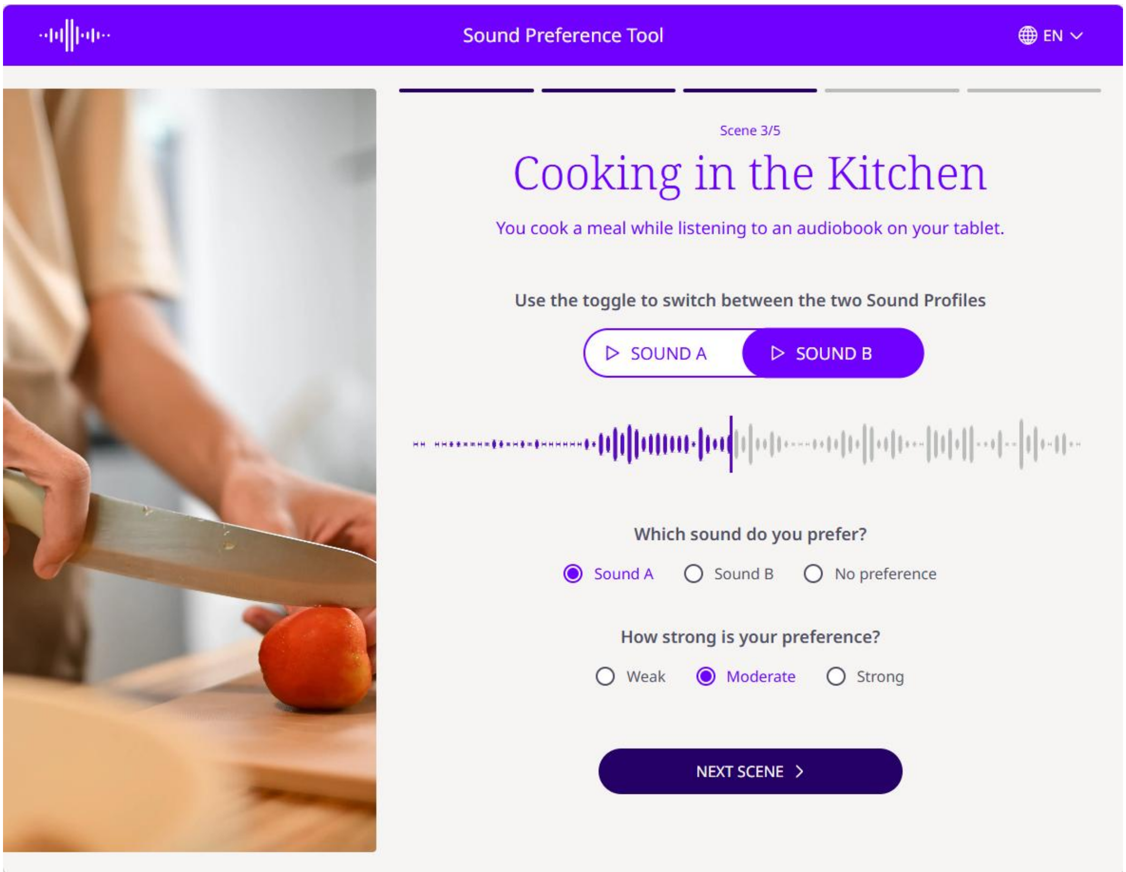
Abstract

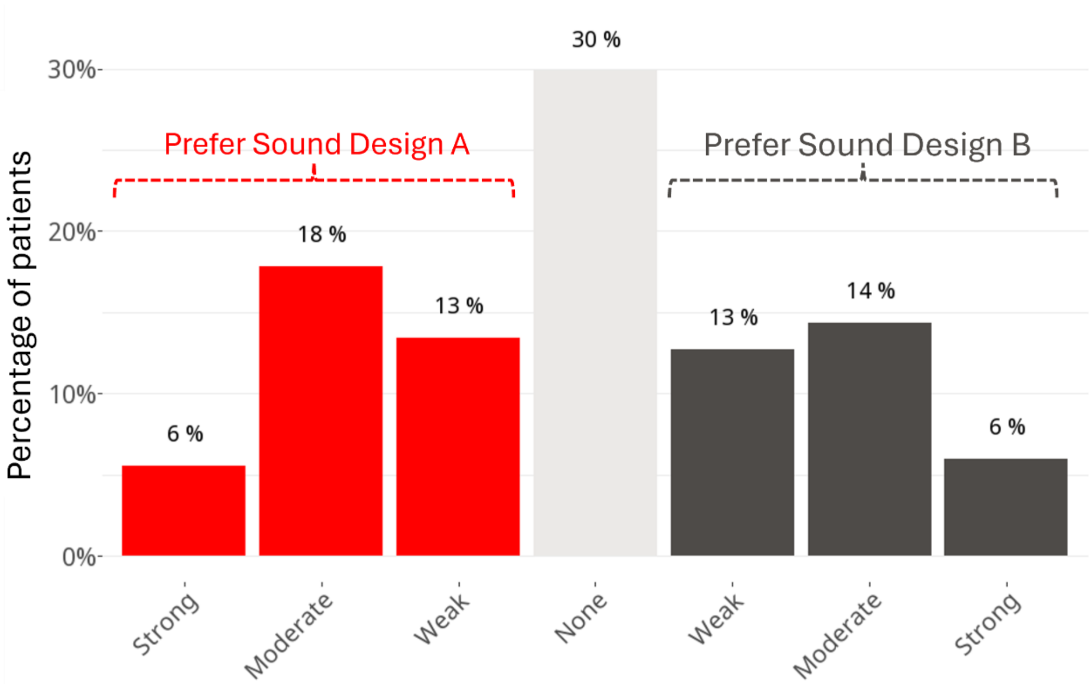
Background. Clinical audiology aims to enable seamless participation in everyday life, including effective communication, enjoyment of music, and awareness of surrounding environments. Central to this goal is the ability to adapt comfortably to hearing aids (HAs). However, achieving an optimal hearing experience is challenging, as no single sound design meets the preferences of all individuals. Previous research on sound preference has shown that, when comparing HAs with two distinct sound designs, about 40% of listeners prefer one of the two designs. This finding led us to investigate the potential clinical benefits of assessing sound preferences in clinics to support HA recommendations.

Methods. We developed the Sound Preference Tool (SPT; https://soundpreference.net), a clinical tool that lets patients compare two HA sound designs that are based on fundamentally different processing architectures. The SPT probes preferences for five everyday situations, asks which sound design was preferred overall, and issues a recommendation. To evaluate clinical benefits, 48 clinicians used the SPT in their clinical workflow to support their HA recommendations. We assess the effect and experience of the SPT by comparing clinical records before and after clinicians implemented it in their workflow.

Results. Preliminary results based on 432 patients confirm previous findings that ~40% of patients have a moderate or strong preference for one or the other sound design. Using the SPT was easy for most patients, but some older patients experienced difficulties. Clinicians reported that the SPT supports more patient-centered conversations, helps decision making, and encourages a broader recommendation approach, reducing single-brand bias. Data collection is ongoing.

Conclusions. The SPT was received positively by both clinicians and patients and adds to existing evidence that a substantial proportion of patients have a strong or moderate preference for a specific sound design.


Sound Preference Tool
EN
Scene 3/5
Cooking in the Kitchen
You cook a meal while listening to an audiobook on your tablet.
Use the toggle to switch between the two Sound Profiles
SOUND A
SOUND B
Which sound do you prefer?
Sound A
Sound B
No preference
How strong is your preference?
Weak
Moderate
Strong
NEXT SCENE


Percentage of patients
30%
20%
10%
0%
Prefer Sound Design A
Prefer Sound Design B
30 %
6 %
18 %
13 %
13 %
14 %
6 %
Strong
Moderate
Weak
None
Weak
Moderate
Strong

Exploration of the Effectiveness of Online Remote Programming for Domestic Cochlear Implant Systems

Ruijie Wang (Shandong Provincial Second People's Hospital); Lei Xu (Shandong Provincial Second People's Hospital); Tianyi Liu (Zhejiang Nurotron Biotechnology co., ltd)*; Shanxian Gao (Zhejiang Nurotron Biotechnology co., ltd); Li Yin (Zhejiang Nurotron Biotechnology co., ltd)

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Podium

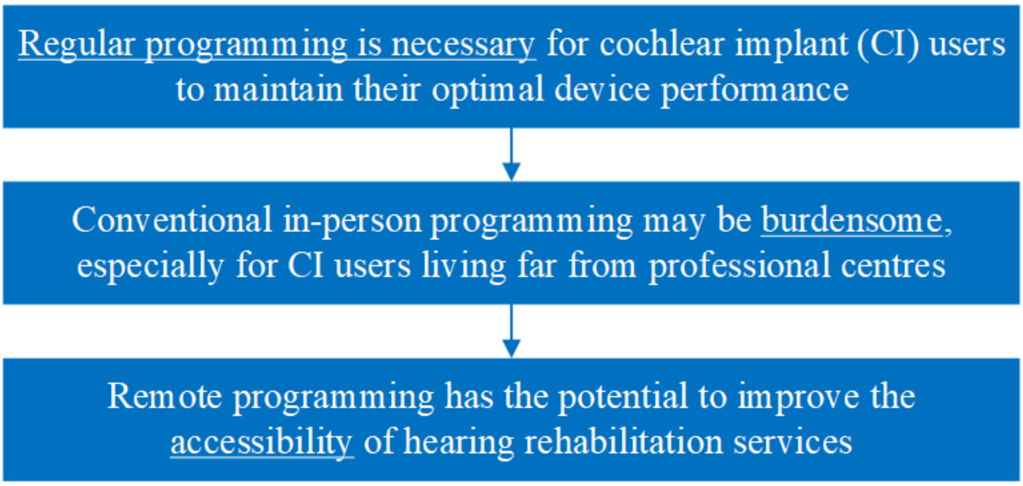
Abstract

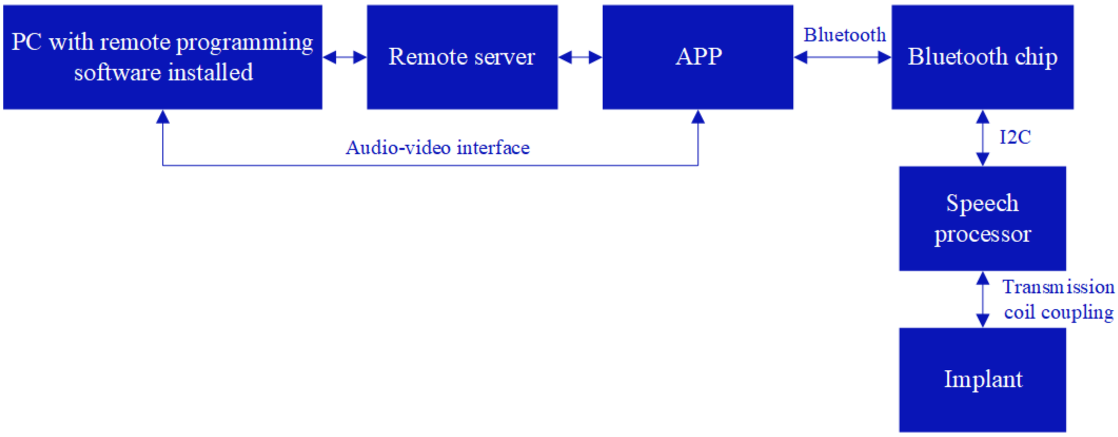
Background. To evaluate the effectiveness of remote programming for domestic cochlear implant (CI) systems and provide clinical trial evidence to promote remote hearing rehabilitation services.

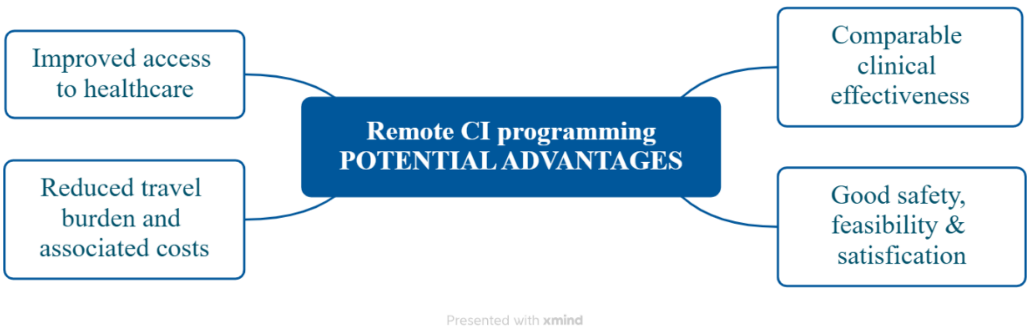
Methods. Twelve subjects implanted with the domestic Nurotron cochlear implant who had been using the device for more than six months (average age 45.58 ± 10.85 years) were enrolled. Each participant underwent one online remote programming session and one in-person traditional programming session in random order. The clinical efficacy of remote programming was assessed using aided hearing thresholds, speech recognition rates, and subjective questionnaires. A non-inferiority test was applied to analyse the difference in outcomes between remote and traditional programming.

Results. The aided hearing thresholds across all frequencies and the speech recognition rates in quiet environments met the predefined non-inferiority margins (Δ = 4/5 dB or 5%). Subjective questionnaire scores indicated good user acceptance, demonstrating that remote programming was non-inferior to traditional programming in both objective and subjective evaluations.

Conclusions. Online remote programming of domestic cochlear implants demonstrated comparable effectiveness to conventional in-person programming for hearing improvement and speech recognition, exhibiting good feasibility and clinical utility. The remote programming function has the potential to enhance the rehabilitation experience for CI users, particularly benefiting those in medically underserved areas or with limited mobility.


Regular programming is necessary for cochlear implant (CI) users to maintain their optimal device performance
Conventional in-person programming may be burdensome, especially for CI users living far from professional centres
Remote programming has the potential to improve the accessibility of hearing rehabilitation services


PC with remote programming software installed
Remote server
APP
Bluetooth
Bluetooth chip
Audio-video interface
I2C
Speech processor
Transmission coil coupling
Implant


Improved access to healthcare
Reduced travel burden and associated costs
Remote CI programming POTENTIAL ADVANTAGES
Comparable clinical effectiveness
Good safety, feasibility & satisfication
Presented with xmind

Loudness scaling with variations in remote mobile hardware: Can we trust the results?

Thomas Schwarz (Carl von Ossietzky Universität Oldenburg)*; Lena Schell-Majoor (Carl von Ossietzky Universität Oldenburg); Birger Kollmeier (Carl von Ossietzky Universität Oldenburg)

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Podium

Abstract

Background. The "Adaptive Categorical Loudness Scaling" (ACALOS) is an established method in audiology to characterize suprathreshold hearing abilities individually, e.g., the residual dynamic range or recruitment effect. Its application for remote testing in audiology with uncalibrated hardware appears attractive, especially if the absolute hearing threshold and a calibration factor can be determined with the same test procedure by employing additional stimulus presentations near the estimated threshold level ("reinforced" ACALOS (rACALOS, Xu et al., 2025)). However, technical limitations of the mobile hardware leading to dynamic range restrictions (nonlinear sound processing, automatic gain control, clipping) may limit the applicability of the method which motivates this study.

Method: This pilot study investigates the impact of different hardware combinations on individual rACALOS data and derived loudness functions when using the Virtual Hearing Clinic (VHC), a modular remote testing platform. Five normal-hearing participants performed monaural tests with a 4 kHz narrow-band-noise stimulus in the lab. Four headphone models (Bluetooth in-ear, wired circumaural headphones) and two playback devices (smartphone, laptop) were combined into eight hardware configurations.

Results. For each hardware configuration, individual loudness functions and threshold estimates are obtained and the intra- and interindividual variance for normal hearing listeners is analyzed and compared to normal hearing reference data.

Conclusions. The magnitude of variance introduced by the use of different hardware significantly affects the feasibility of rACALOS in remote testing. The results will enable an estimation of expected accuracy and reliability for loudness scaling data obtained with the VHC.

References: Xu, C., & Kollmeier, B. (2025). Calibration offset estimation in mobile hearing tests via categorical loudness scaling. arXiv preprint arXiv:2508.14824.

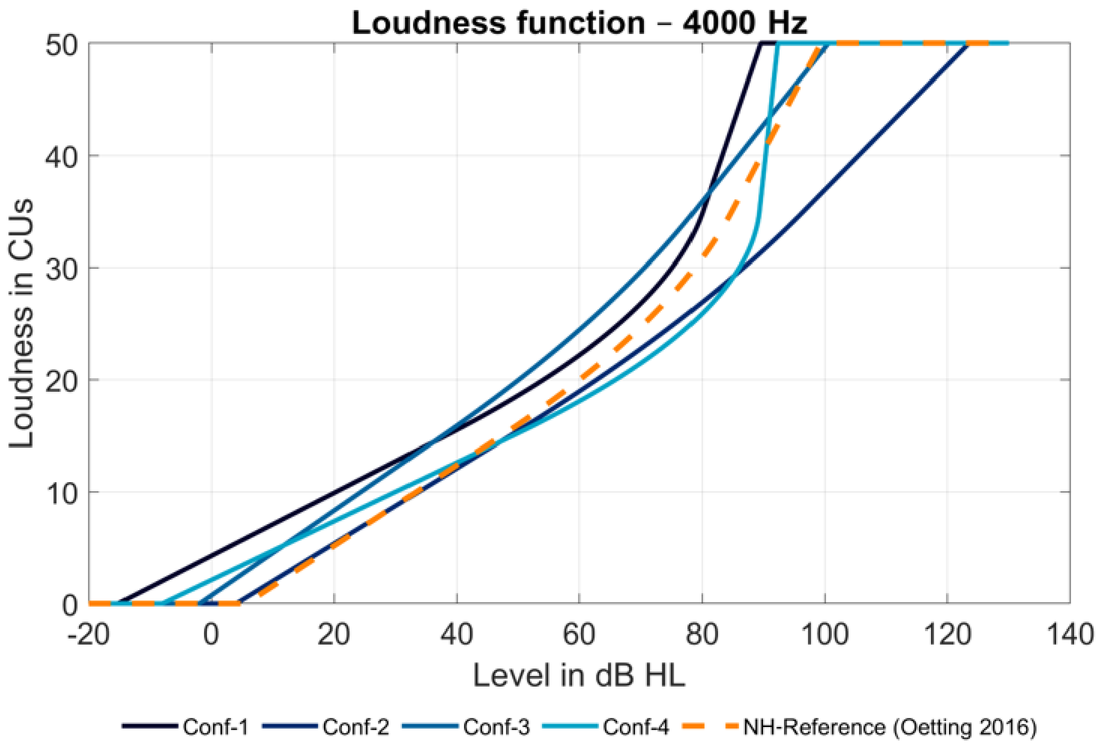

Figure 1: Categorical loudness functions for the same normal hearing subject obtained with different mobile hardware combinations and normal-hearing reference (preliminary data for demonstration purposes).

Knowledge and Attitudes Toward Tele-Audiology among Hearing Care Professionals

Şule Çekiç (Ankara Yildirim Beyazit University, Department of Audiology); Hande Çakıroğlu (Ankara Yildirim Beyazit University, Institute of Health Sciences)*; David Tomé (Poly Technich Instutute of Porto); Nil Gizem Eyrekkaya (Ankara Yildirim Beyazit University, Institute of Health Sciences)

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Poster

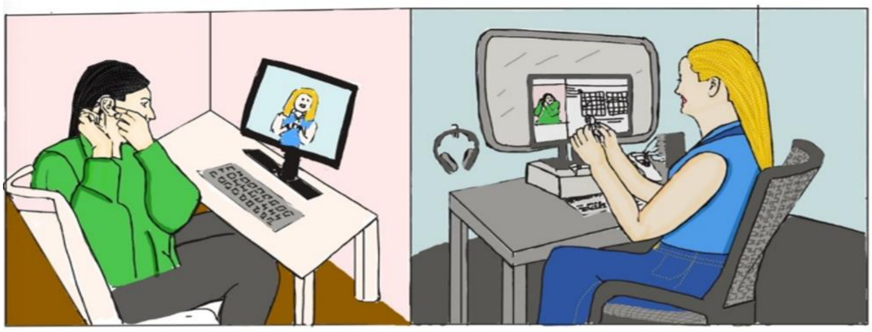
Abstract

Background. Hearing loss is a growing global health concern. Tele-audiology can improve access to care through remote services, yet its implementation remains limited. In Türkiye, research on practitioners' knowledge and attitudes is insufficient.

Methods. This prospective, quantitative survey included 75 practitioners (18–65 years) from 55 hearing aid centers. Data were collected via digital platforms and WhatsApp using a 27-item questionnaire administered through Google Forms. Analyses were conducted with IBM SPSS Statistics.

Results. Approximately 54.7% of participants were familiar with tele-audiology. Its main use was hearing aid fitting and follow-up (97.6%). No significant associations was found between knowledge and gender (p=0.701) or professional title (p=0.840). Among prior users, 53.3% found it useful, and prior experience was significantly associated with perceived usefulness (p=0.001). Additionally, 60% requested further training and 86.7% suggested improvements in current systems; satisfaction was significantly associated with the desire for improvement (p=0.035). Most users reported no negative feedback from patients.

Conclusions. Tele-audiology in Türkiye remains underutilized despite its potential to expand access to hearing care. Although practitioners are generally open to its use, applications are largely limited to device-related services, indicating a gap between awareness and broader clinical implementation. These findings highlight the need for targeted training, stronger infrastructure, and standardized service models to support wider integration into routine practice.



Developing RNID’s Vision for the Future of Hearing Healthcare

Crystal Rolfe (Royal National Institute for Deaf People); Lola Russell (Royal National Institute for Deaf People)*; Franki Oliver (Royal National Institute for Deaf People); Alastair Moore (Royal National Institute for Deaf People); Victoria Boelman (Royal National Institute for Deaf People)

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Poster

Abstract

Background. Across the UK, it is estimated that over 10 million people could benefit from hearing aids, but only 3 million people use them. When people try to get help with their hearing, they struggle with confusing processes, limited appointments, and inflexible support options. Untreated hearing loss has significant negative consequences for social connection, loneliness, mental health, and health-related quality of life. Improving access to hearing healthcare is a critical public health issue. RNID has been exploring how technology could transform adult hearing healthcare by improving efficiency and enabling services to better meet people’s needs.

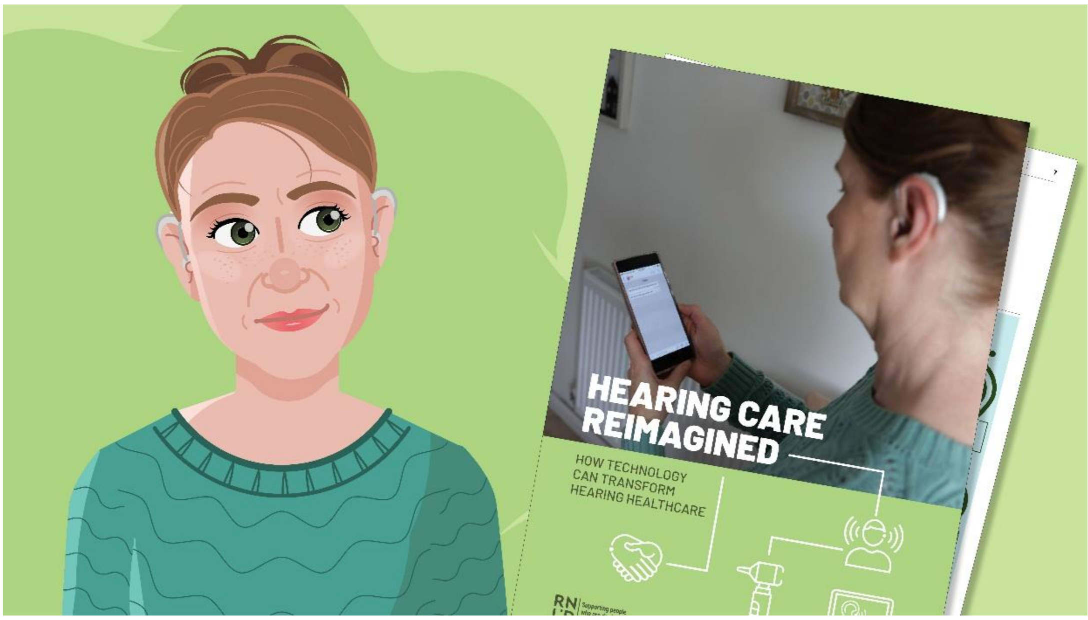
Methods. We used a mixed methods approach including surveys, interviews, workshops, and focus groups to capture perspectives from healthcare professionals, academics, technologists, and people with lived experience.

Results. People with lived experience expressed a strong need for more accessible, flexible, and person-centred service. Key themes included:

- Lack of routine hearing checks and public awareness
- Confusing and complex referral processes
- Interest in technologies that support easier and more flexible ways to get support
- A strong desire for self-management tools and accessible information

Professionals highlighted technologies tested in research environments, healthcare services for other long-term conditions, or in other countries.

Conclusions. The findings identified four key opportunities for technology to improve services: identifying early signs of hearing loss; support quicker diagnosis of hearing loss; make it easier for people to get support; empower people to manage their hearing health. These findings informed RNID’s “Hearing Care Reimagined Report.” Beyond individual technologies, there is also the potential to harness population health data to create meaningful benefits for people, services, and the health system.


HEARING CARE
REIMAGINED
HOW TECHNOLOGY
CAN TRANSFORM
HEARING HEALTHCARE

Comparative Study of Auditory Gestalt Perception Using AI-Generated Clear Speech in Sports and Non-Sports Trainees

Sneha Chandra Sekar (SRM Institute of Science and Technology)*

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Poster

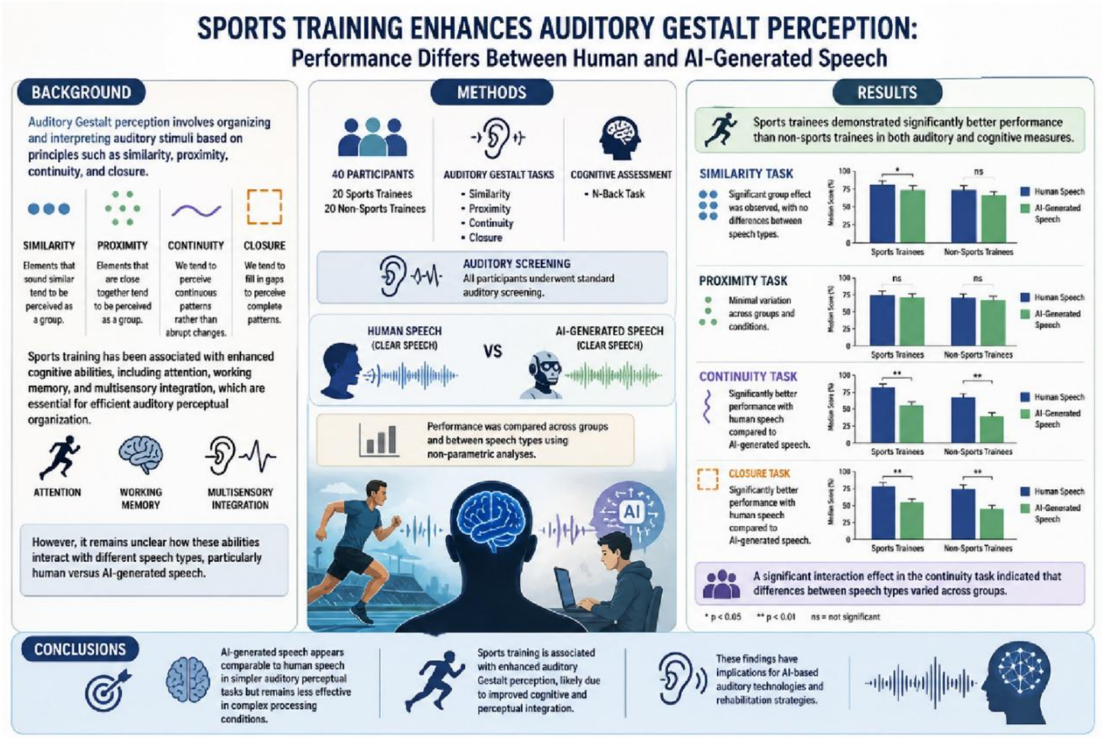
Abstract

Background. Auditory Gestalt perception involves organizing and interpreting auditory stimuli based on principles such as similarity, proximity, continuity, and closure. Sports training has been associated with enhanced cognitive abilities, including attention, working memory, and multisensory integration, which are essential for efficient auditory perceptual organization. However, it remains unclear how these abilities interact with different speech types, particularly human versus AI-generated speech.

Methods. Forty participants (20 sports trainees and 20 non-sports trainees) were included in this study. All participants underwent auditory Gestalt tasks assessing similarity, proximity, continuity, and closure, along with cognitive assessment using the N-Back task and standard auditory screening. Performance was compared across groups and between human speech and AI-generated clear speech conditions using non-parametric analyses.

Results. Sports trainees demonstrated significantly better performance than non-sports trainees in both auditory and cognitive measures. In the similarity task, a significant group effect was observed, with no differences between speech types. The proximity task showed minimal variation across groups and conditions. In contrast, continuity and closure tasks revealed significantly better performance with human speech compared to AI-generated speech. A significant interaction effect in the continuity task indicated that differences between speech types varied across groups.

Conclusions. AI-generated speech appears comparable to human speech in simpler auditory perceptual tasks but remains less effective in complex processing conditions. Sports training is associated with enhanced auditory Gestalt perception, likely due to improved cognitive and perceptual integration. These findings have implications for AI-based auditory technologies and rehabilitation strategies.



SPORTS TRAINING ENHANCES AUDITORY GESTALT PERCEPTION:
Performance Differs Between Human and AI-Generated Speech
BACKGROUND
Auditory Gestalt perception involves organizing and interpreting auditory stimuli based on principles such as similarity, proximity, continuity, and closure.
SIMILARITY
Elements that sound similar tend to be perceived as a group.
PROXIMITY
Elements that are close together tend to be perceived as a group.
CONTINUITY
We tend to perceive continuous patterns rather than abrupt changes.
CLOSURE
We tend to fill in gaps to perceive complete patterns.
Sports training has been associated with enhanced cognitive abilities, including attention, working memory, and multisensory integration, which are essential for efficient auditory perceptual organization.
ATTENTION
WORKING MEMORY
MULTISENSORY INTEGRATION
However, it remains unclear how these abilities interact with different speech types, particularly human versus AI-generated speech.
METHODS
40 PARTICIPANTS
20 Sports Trainees
20 Non-Sports Trainees
AUDITORY GESTALT TASKS
• Similarity
• Proximity
• Continuity
• Closure
COGNITIVE ASSESSMENT
• N-Back Task
AUDITORY SCREENING
All participants underwent standard auditory screening.
HUMAN SPEECH (CLEAR SPEECH)
VS
AI-GENERATED SPEECH (CLEAR SPEECH)
Performance was compared across groups and between speech types using non-parametric analyses.
AI
RESULTS
Sports trainees demonstrated significantly better performance than non-sports trainees in both auditory and cognitive measures.
SIMILARITY TASK
Significant group effect was observed, with no differences between speech types.
PROXIMITY TASK
Minimal variation across groups and conditions.
CONTINUITY TASK
Significantly better performance with human speech compared to AI-generated speech.
CLOSURE TASK
Significantly better performance with human speech compared to AI-generated speech.
Sports Trainees
Non-Sports Trainees
Human Speech
AI-Generated Speech
A significant interaction effect in the continuity task indicated that differences between speech types varied across groups.
* p < 0.05 ** p < 0.01 ns = not significant
CONCLUSIONS
AI-generated speech appears comparable to human speech in simpler auditory perceptual tasks but remains less effective in complex processing conditions.
Sports training is associated with enhanced auditory Gestalt perception, likely due to improved cognitive and perceptual integration.
These findings have implications for AI-based auditory technologies and rehabilitation strategies.

Session 5.A. Cochlear Implants: Challenges and Prospects

Advances in signal processing, computational models, neural interfaces, and personalised stimulation strategies.

Chaired by Prof. Tania Hanekom.

Neural Response-Driven Optimization of Cochlear Implant Stimulation Using Differentiable Auditory Models

Julie Van Heghe (Ghent University)*; Sarah Verhulst (Ghent University); Guy Torfs (Ghent University)

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Featured talk

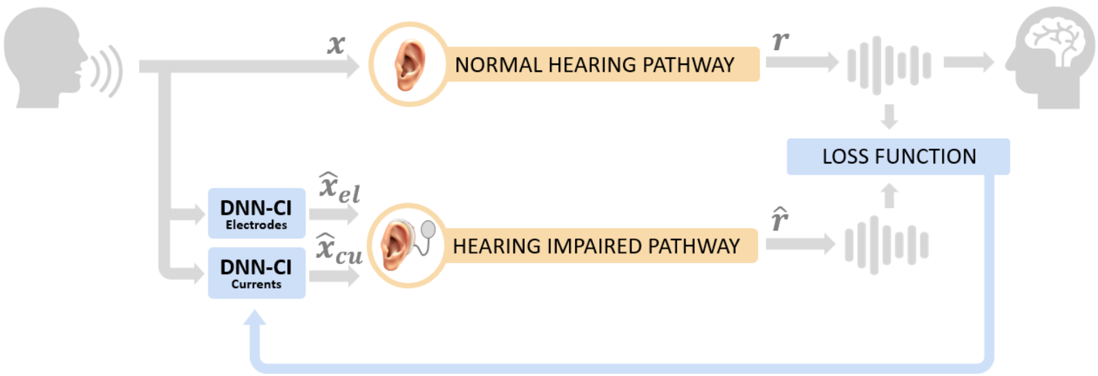
Abstract

Background. Cochlear implants (CIs) restore hearing by electrically stimulating the auditory nerve, yet current sound-processing strategies produce neural representations that differ substantially from those in normal hearing. This mismatch limits perceptual performance, particularly in complex listening environments. Approaches that directly target neural response similarity may help bridge this gap.

Methods. We propose a closed-loop optimization framework that combines computational auditory models with deep learning to design CI stimulation strategies. The system simulates both normal-hearing and cochlear-implant auditory pathways, generating spiral ganglion neuron responses to the same acoustic input. Two neural networks replace the conventional sound-coding stage and jointly learn electrode activation patterns and current amplitudes through end-to-end training. Optimization minimizes the difference between normal-hearing and CI-evoked neural responses using differentiable approximations of auditory periphery models.

Results. In a proof-of-concept evaluation using speech stimuli, the learned stimulation strategy substantially reduced the mismatch between normal-hearing and CI neural response envelopes compared to the conventional Advanced Combination Encoder (ACE), achieving a 6.6-fold reduction in envelope error. While the current implementation does not yet fully account for all practical constraints of clinical CI devices, these results indicate improved alignment of simulated neural activity.

Conclusions. These findings demonstrate the feasibility of closed-loop, model-based optimization for cochlear implant processing. By integrating auditory neuroscience models with machine learning, the proposed framework offers a promising direction for developing next-generation CI strategies that more closely reproduce neural responses associated with normal hearing.


x
NORMAL HEARING PATHWAY
r
LOSS FUNCTION
DNN-CI
Electrodes
DNN-CI
Currents
$\hat{x}_{el}$
$\hat{x}_{cu}$
HEARING IMPAIRED PATHWAY
$\hat{r}$

Comparative Study of AutoNRM Intelligent Mapping and Manual Mapping in Nurotron Cochlear Implant Recipients

Yipeng LIN (Zhejiang Nurotron Biotechnology Co. Ltd.)*; Yong Tao (West China Hospital); Huilin Yin (West China Hospital); Shanxian Gao (Zhejiang Nurotron Biotechnology Co. Ltd.); Li Yin (Zhejiang Nurotron Biotechnology Co. Ltd.); Gang Li (West China Hospital)

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Podium

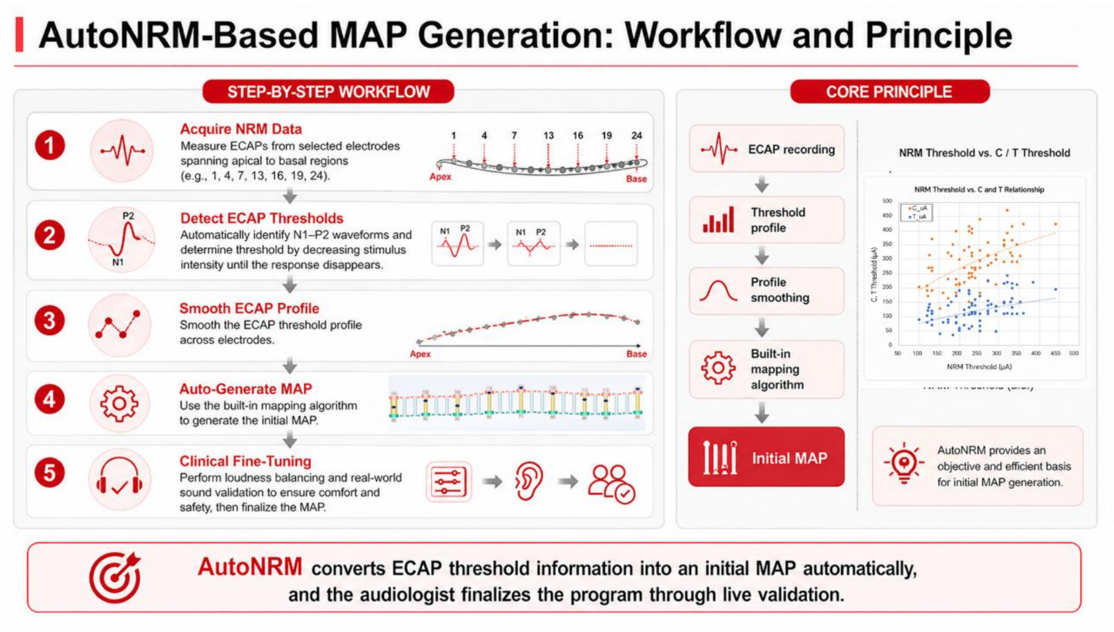
Abstract

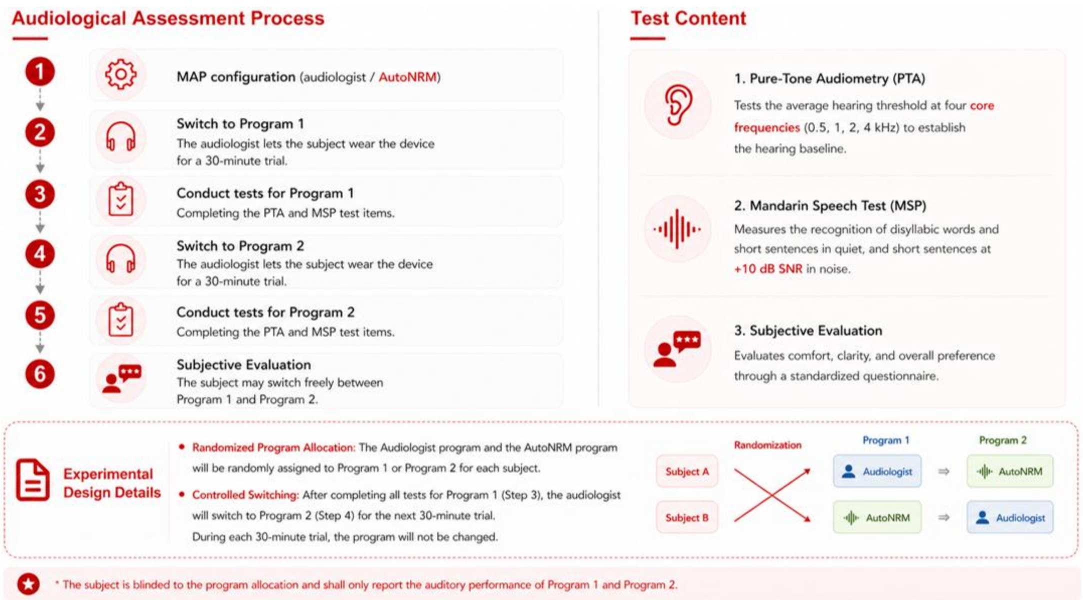
Background. This study evaluated the safety and clinical effectiveness of AutoNRM, an automatic fitting system integrated into Nurotron's Nurosound platform. AutoNRM automatically measures electrically evoked compound action potential (ECAP) thresholds and generates an ECAP-based initial fitting map, which is further fine-tuned by audiologists in live mode to create a finalized LiveECAPMAP. Outcomes were compared with conventional behaviorally based LiveBurstMAPs.

Methods. Twenty cochlear implant users without cochlear malformations were enrolled (6 males, 14 females; mean age 36.05 ± 14.85 years). ECAP thresholds were recorded from electrodes 1, 4, 7, 13, 16, 19, and 24. AutoNRM generated ECAPMAPs based on ECAP thresholds and AGF curves, followed by audiologist-guided loudness balancing and real-world listening checks. After a 30-minute adaptation period for each MAP, participants completed aided hearing threshold testing at 0.5, 1, 2, and 4 kHz, Mandarin Speech Perception tests, including disyllabic words in quiet, sentences in quiet, and sentences in noise at 10 dB SNR, and an 11-point subjective questionnaire evaluating comfort, intelligibility, clarity, and overall preference. A double-blind design was used.

Results. Aided thresholds were comparable between LiveECAPMAPs and LiveBurstMAPs (31.81 ± 3.59 vs. 32.36 ± 4.62 dB HL; paired t-test, $p > 0.05$). No significant differences were observed in disyllabic word recognition in quiet (69.66 ± 21.84% vs. 73.36 ± 19.96%), sentence recognition in quiet (66.59 ± 25.05% vs. 67.31 ± 22.12%), or sentence recognition in noise (54.56 ± 33.81% vs. 55.33 ± 33.65%; all $p > 0.05$). Subjective ratings also showed no significant differences between MAP types.

Conclusions. AutoNRM enables reliable ECAP-based automatic generation of initial cochlear implant fitting maps. Its clinical outcomes are comparable to conventional psychophysical fitting, supporting its safety, reliability, and efficiency for initial fitting in adult CI users.


AutoNRM-Based MAP Generation: Workflow and Principle
STEP-BY-STEP WORKFLOW
1 Acquire NRM Data
Measure ECAPs from selected electrodes spanning apical to basal regions (e.g., 1, 4, 7, 13, 16, 19, 24).
1 4 7 13 16 19 24
Apex
Base
2 Detect ECAP Thresholds
Automatically identify N1–P2 waveforms and determine threshold by decreasing stimulus intensity until the response disappears.
P2
N1
N1 P2
N1 P2
3 Smooth ECAP Profile
Smooth the ECAP threshold profile across electrodes.
Apex
Base
4 Auto-Generate MAP
Use the built-in mapping algorithm to generate the initial MAP.
5 Clinical Fine-Tuning
Perform loudness balancing and real-world sound validation to ensure comfort and safety, then finalize the MAP.
CORE PRINCIPLE
ECAP recording
Threshold profile
Profile smoothing
Built-in mapping algorithm
Initial MAP
NRM Threshold vs. C / T Threshold
NRM Threshold vs. C and T Relationship
NRM Threshold (μA)
AutoNRM provides an objective and efficient basis for initial MAP generation.
AutoNRM converts ECAP threshold information into an initial MAP automatically, and the audiologist finalizes the program through live validation.


Audiological Assessment Process
1 MAP configuration (audiologist / AutoNRM)
2 Switch to Program 1
The audiologist lets the subject wear the device for a 30-minute trial.
3 Conduct tests for Program 1
Completing the PTA and MSP test items.
4 Switch to Program 2
The audiologist lets the subject wear the device for a 30-minute trial.
5 Conduct tests for Program 2
Completing the PTA and MSP test items.
6 Subjective Evaluation
The subject may switch freely between Program 1 and Program 2.
Test Content
1. Pure-Tone Audiometry (PTA)
Tests the average hearing threshold at four core frequencies (0.5, 1, 2, 4 kHz) to establish the hearing baseline.
2. Mandarin Speech Test (MSP)
Measures the recognition of disyllabic words and short sentences in quiet, and short sentences at +10 dB SNR in noise.
3. Subjective Evaluation
Evaluates comfort, clarity, and overall preference through a standardized questionnaire.
Experimental Design Details
• Randomized Program Allocation: The Audiologist program and the AutoNRM program will be randomly assigned to Program 1 or Program 2 for each subject.
• Controlled Switching: After completing all tests for Program 1 (Step 3), the audiologist will switch to Program 2 (Step 4) for the next 30-minute trial.
During each 30-minute trial, the program will not be changed.
Randomization
Program 1
Program 2
Subject A
Subject B
Audiologist
AutoNRM
AutoNRM
Audiologist
* The subject is blinded to the program allocation and shall only report the auditory performance of Program 1 and Program 2.

Prediction of Speech Outcomes with Cochlear Implants: Limitations and Opportunities for Clinical Practice

Marta Campi (Deep Hearing Lab, University of Zurich)*; Alexander Huber (University Hospital Zurich, University of Zurich); Tobias Goehring (Deep Hearing Lab, University of Zurich)

marta.campi.11@gmail.com

Podium

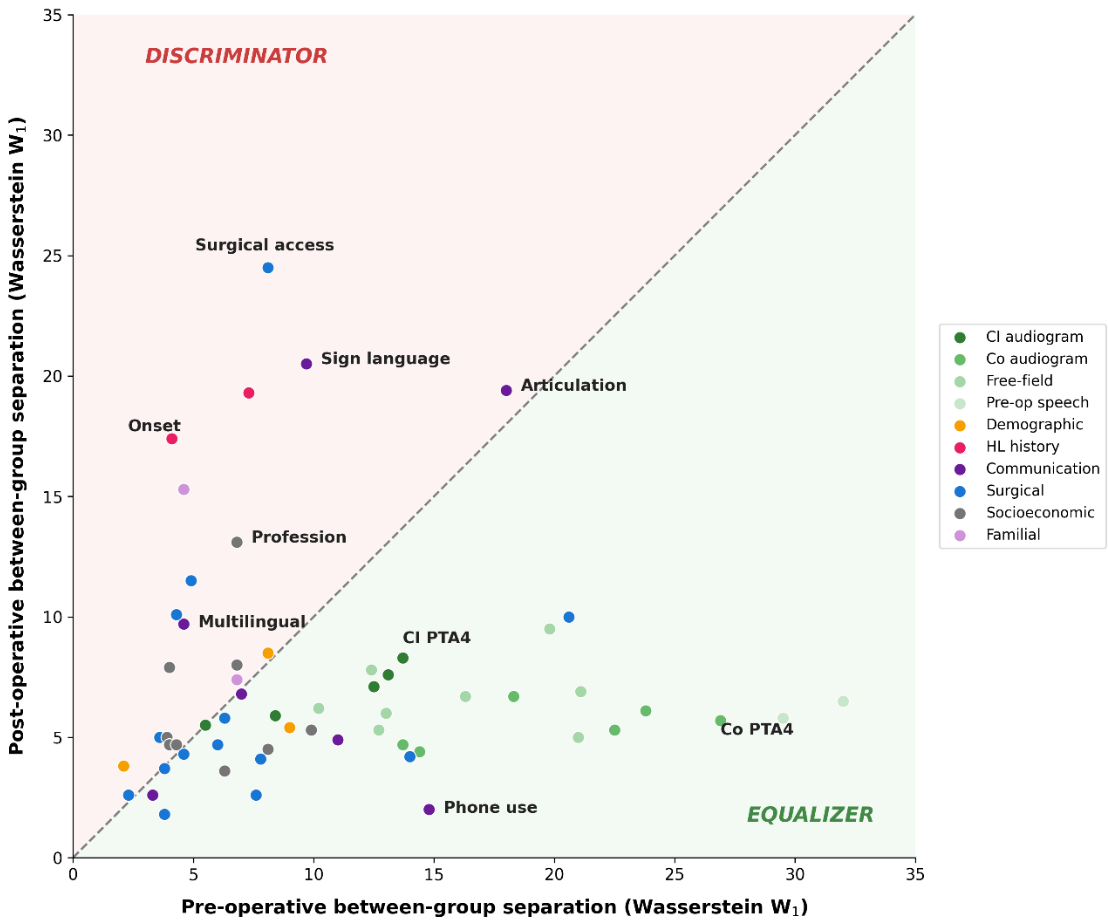
Abstract

Background. The prediction of individual speech outcomes after cochlear implantation remains challenging, with reported explained variances of R2 = 0.07 to 0.27 and prediction errors too large for clinical counselling. Whether this reflects insufficient modelling capability or a fundamental limit of the available predictor space is unknown. Here we compare statistical models across prediction tasks and discuss their clinical relevance.

Methods. We performed a retrospective analysis of a single-centre adult cohort (538 adults) from the Swiss National Cochlear Implant Database using linear models, ensemble methods, and a state-of-the-art foundation model for tabular data, with varying feature sets and imputation strategies under cross-validation. We compared continuous score prediction, binary classification (poor outcome, decline), and unsupervised profiling of post-operative speech patterns across five speech tests.

Results. No modelling approach achieved clinically useful prediction of individual post-operative scores (cross-validated R2 below 0.03). Patients across hearing-loss severity converge to overlapping post-operative ranges, limiting the variance available for regression. Binary classification - identifying patients performing below a clinical threshold or declining after implantation - proved more successful (AUC = 0.67 and 0.77 respectively), with onset type, articulation, contralateral hearing, and duration of hearing loss as consistent predictors. Unsupervised clustering identified distinct performance subgroups, including patients with atypical patterns across speech dimensions, but these could not be predicted from pre-operative data alone.

Conclusions. The limitation appears informational rather than algorithmic. We propose alternatives to score prediction: pre-operative risk screening, post-operative profiling to guide rehabilitation, and early post-operative monitoring to identify patients requiring intervention.



DISCRIMINATOR
EQUALIZER
Surgical access
Sign language
Articulation
Onset
Profession
Multilingual
CI PTA4
Co PTA4
Phone use
CI audiogram
Co audiogram
Free-field
Pre-op speech
Demographic
HL history
Communication
Surgical
Socioeconomic
Familial
Pre-operative between-group separation (Wasserstein W_1)
Post-operative between-group separation (Wasserstein W_1)

From Clinic to Reality: Integrating Sound Field Testing and Hearing Quality Measures in Cochlear Implant Users

Marta Álvarez-Cendrero (Hospital Universitario La Merced)*; Manuel Lazo- Maestre (Hospital Universitario Virgen Macarena); María A. Callejón-Leblic (Hospital Universitario Virgen Macarena)

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Podium

Abstract

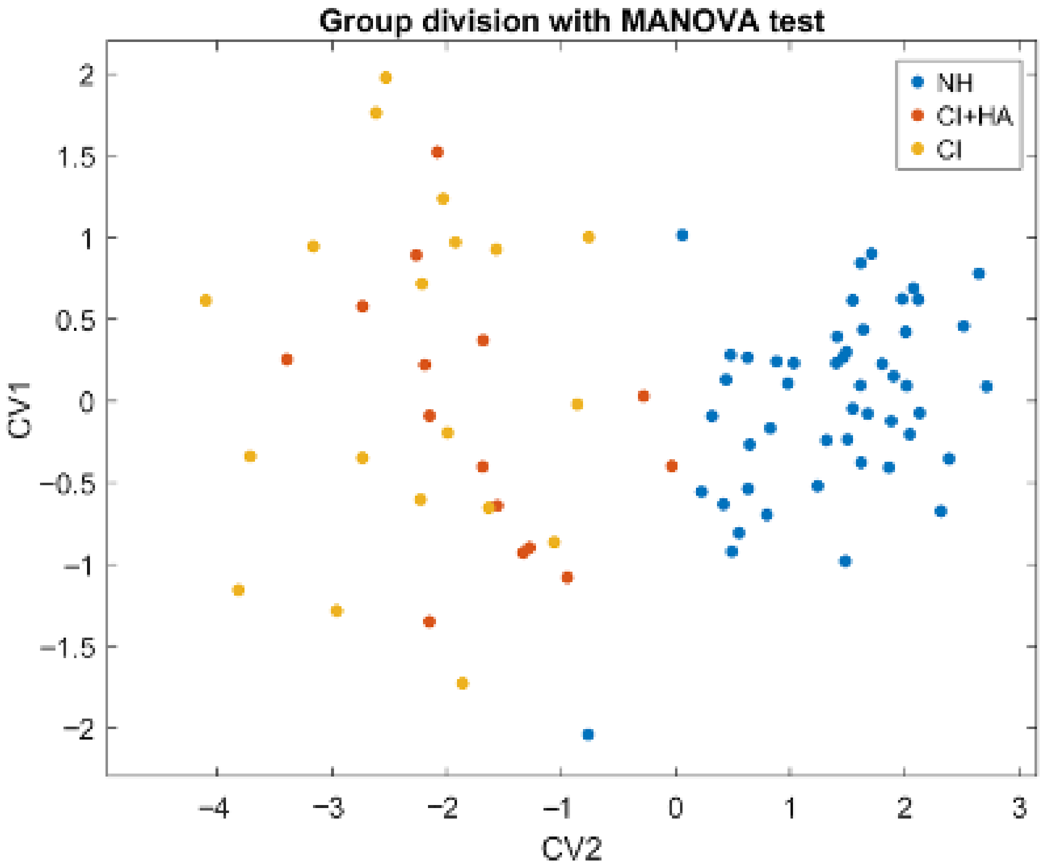
Background. Conventional audiological assessments often fail to capture the real-world listening difficulties experienced by cochlear implant (CI) users, particularly in complex acoustic environments. Limitations in speech recognition and sound localization may persist despite clinically acceptable audiometric outcomes.

Objective: This study aims to investigate how acoustic environment complexity affects speech recognition and sound localization in CI users compared to normal-hearing (NH) individuals, and to evaluate subjective auditory performance using validated questionnaires.

Methods. Thirty-four unilateral CI users and fifty NH participants were tested in a sound-treated room equipped with a 360° eight-speaker array. Measures included pure-tone average (PTA), word recognition scores (WRS) in quiet and in noise (sea, traffic, cafeteria), sound localization ability, and the SSQ-12 questionnaire. Relationships between objective and subjective outcomes were analysed using correlation, regression, and MANOVA.

Results. CI users showed significantly poorer WRS in both quiet and noisy conditions, reduced localization accuracy, and lower SSQ-12 scores compared to NH participants. Performance declined with increasing noise complexity. Pre-implant PTA emerged as the strongest predictor of post-implant speech recognition. No significant differences were found between unilateral and bimodal CI users. SSQ-12 scores were significantly associated with objective measures, supporting its clinical utility.

Conclusions. CI users continue to experience deficits in realistic listening environments that are not captured by standard audiometry. Combining sound field assessments with validated auditory questionnaires may offer a more comprehensive evaluation to inform individualized rehabilitation and device optimization.


Group division with MANOVA test
NH
CI+HA
CI
CV1
CV2
2
1.5
1
0.5
0
−0.5
−1
−1.5
−2
−4
−3
−2
−1
0
1
2
3

Clinical Validation of the Web-based Platform AUDITO for Cochlear Implant Patients

Andreas Markidis (University of Cambridge)*; James R. Tysome (University of Cambridge); Tobias Goehring (University of Cambridge); Tsvetemira Koleva (University of Cambridge); Susan Eitutis (University of Cambridge)

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Podium

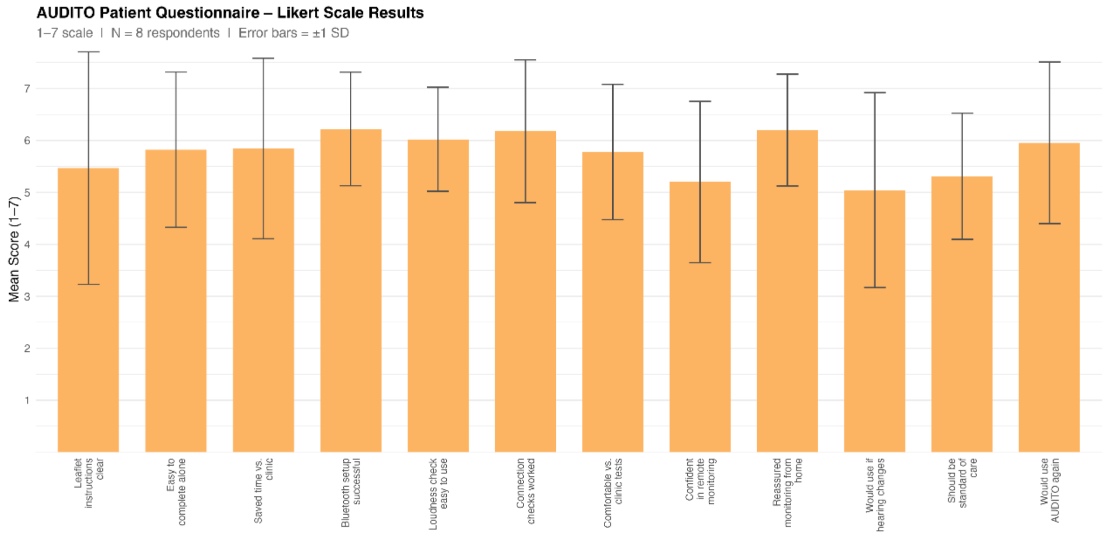
Abstract

Background. Cochlear Implants (CI) require lifelong specialist follow-up, yet growing patient numbers are outpacing clinic capacity. Telehealth has demonstrated reliable remote assessment outcomes, but existing tools are manufacturer specific. AUDITO is the first brand-agnostic web-based platform delivering controlled auditory tests via Bluetooth streaming. Clinical validation is needed to confirm equivalence with standard in-clinic testing.

Methods. Adult CI users attending routine annual review were recruited. Participants completed BKB sentence and AB word list tests in both a calibrated in-clinic booth and independently at home via AUDITO. Response recordings were resampled to 16kHZ in MATLAB prior to scoring. A structured questionnaire captured usability, technical feasibility, acceptability and clinical comparison across four domains. Paired statistical analysis was conducted in R.

Results. Data collection is ongoing. Preliminary findings will report speech perception scores across in-clinic and remote AUDITO conditions, alongside patient-reported outcomes from the usability questionnaire. Statistical comparisons and full results will be reported upon completion.

Conclusions. AUDITO represents a promising step toward decentralising CI care. Clinical Validation of this platform could provide a scalable, brand-agnostic solution for remote CI monitoring across the UK and internationally, reducing patient burden and alleviating clinical workload. Wider adoption would support improved access to specialist audiology care regardless of geography, offering a sustainable model for the growing global CI population.


AUDITO Patient Questionnaire – Likert Scale Results
1–7 scale | N = 8 respondents | Error bars = ±1 SD
Mean Score (1–7)
1
2
3
4
5
6
7
Leaflet instructions clear
Easy to complete alone
Saved time vs. clinic
Bluetooth setup successful
Loudness check easy to use
Connection checks worked
Comfortable vs. clinic tests
Confident in remote monitoring
Reassured monitoring from home
Would use if hearing changes
Should be standard of care
Would use AUDITO again

A combined Computational Model of Hair Cell and Auditory Nerve Responses to Acoustic and Electric Stimulation

Roman Stracke (Hearing and Neuro Technology Group, Dept. of Microelectronics, Universitat Autònoma de Barcelona; Auditory Prostethic Group, Dept. of Otorhinolaryngology, Hannover Medical School), Daniel Kipping (Hearing and Neuro Technology Group, Dept. of Microelectronics, Universitat Autònoma de Barcelona; Cluster of Excellence “Hearing4All”); Franklin Alvarez (Hearing and Neuro Technology Group, Dept. of Microelectronics, Universitat Autònoma de Barcelona; Cluster of Excellence “Hearing4All”), Yixuan Zhang (Hearing and Neuro Technology Group, Dept. of Microelectronics, Universitat Autònoma de Barcelona; Cluster of Excellence “Hearing4All”), Waldo Nogueira (Hearing and Neuro Technology Group, Dept. of Microelectronics, Universitat Autònoma de Barcelona; Auditory Prostethic Group, Dept. of Otorhinolaryngology, Hannover Medical School; Cluster of Excellence “Hearing4All”; Catalan Institution for Research and Advanced Studies).

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Podium

Abstract

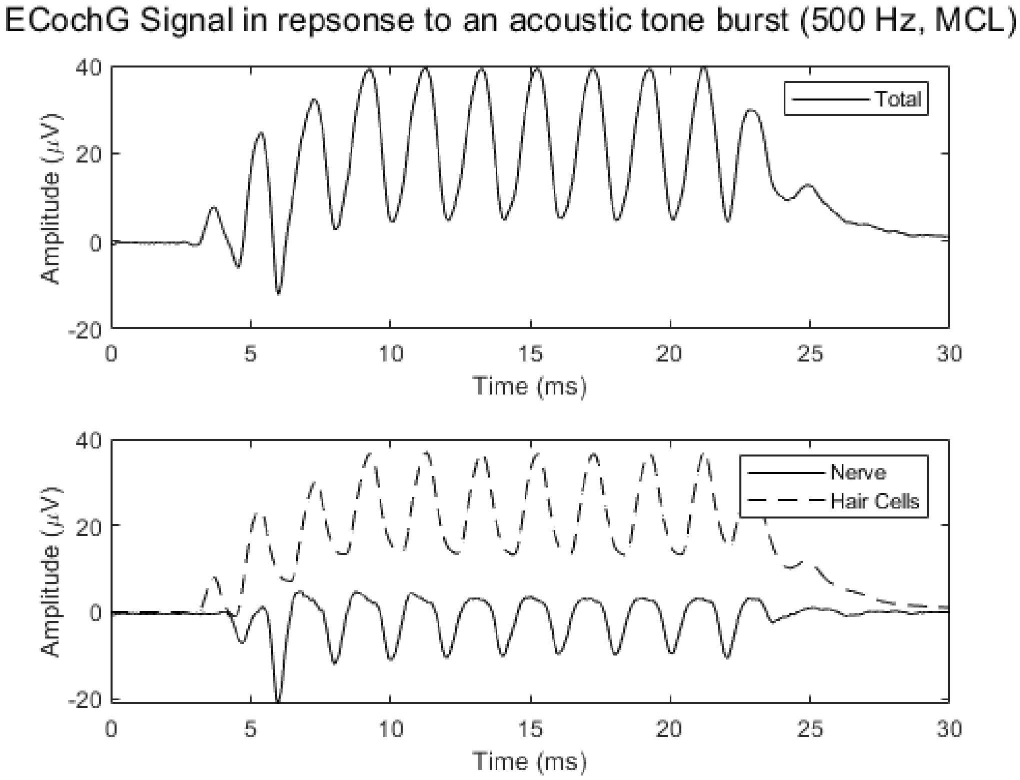
Background. Electrocochleography (ECochG) refers to the recording of electrical potentials from the cochlea and auditory nerve, which enables valuable insights into the auditory periphery. We propose a modeling framework for estimating intracochlear ECochG recordings from hair cells and nerve fibers in response to acoustic and electric stimulation.

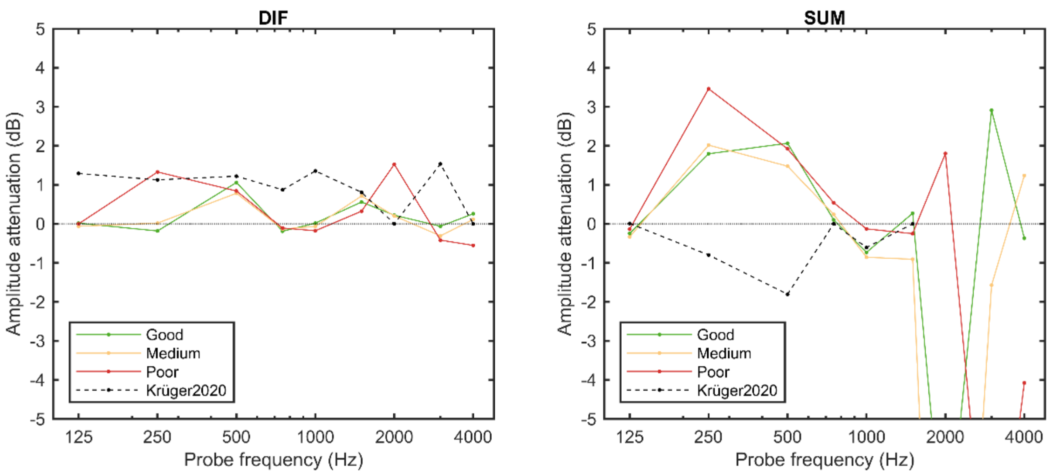
Methods. The framework is constructed by integrating several existing phenomenological and biophysical models of key subsystems of the auditory periphery to individually estimate hair cell and nerve fiber contributions. We conducted two experiments for acoustic-only and electric-acoustic stimulation respectively, in which estimated responses and experimental data were compared in terms of waveform morphology, amplitude growth functions, and electric-acoustic interaction.

Results. Modeled responses for acoustic-only stimulation yielded waveforms and amplitude growth functions in good agreement with experimental data. Modeled waveforms show typical components as cochlear mirophonics, auditory nerve neurophonics, summating potentials, and N1 onset peaks. Electric-acoustic stimulation yielded electric masking, but with lower attenuation when compared to the experiment.

Conclusion: The results suggest a more in-depth examination of the phase relationships of individual ECochG components, particularly the outer hair cells. Further, this work provides a framework for ECochG responses from hair cells and nerve fibers in the context of electric-acoustic stimulation, facilitates the interpretation

of ECochG responses, and adds to the discourse of underlying mechanisms in electric-acoustic interaction.





Longitudinal Analysis of Cochlear Implant Stimulation Levels in the First Year: Optimizing Adult Fitting

Maryam Hussain (Erasmus MC)*; Hylke F.E. van der Toom (Erasmus MC); Mats Voogelaar (Erasmus MC); Bernd Kremer (Erasmus MC); Jantien L. Vroegop (Erasmus MC)

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Poster

Abstract

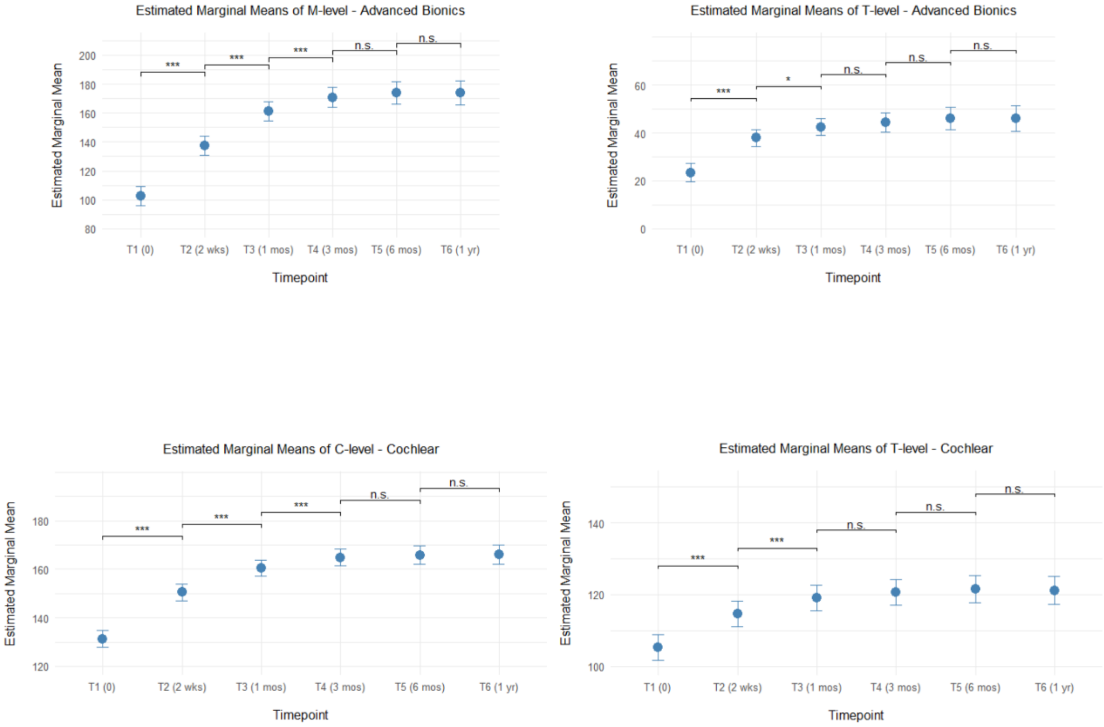
Background. There is no standardized fitting protocol regarding the amount and timing of fitting sessions during CI rehabilitation. Moreover, fitting sessions can be time consuming, contributing to the increasing workload faced by CI centers. In order to optimize the current rehabilitation process, the present study aimed to evaluate the current fitting strategy within the Erasmus MC. The main objective of this study was to investigate the evolution in stimulation levels (C-/M- and T- levels) in adult CI users from both Advanced Bionics™ and Cochlear™, and to assess whether this trajectory differs between the two brands.

Methods. This retrospective study was conducted at the Erasmus University Medical Center in Rotterdam, the Netherlands. A total of 183 post-lingually deaf adults were included with a unilateral CI from either Advanced Bionics™ or Cochlear™. Stimulation levels (C-, M- and T- levels) were collected from these adults for six fitting sessions during the first year post-implantation. Linear mixed-effects models were used to analyze longitudinal changes.

Results. Our findings were comparable for both Advanced Bionics™ and Cochlear™ implants users. Mean M/C and T levels changed over time, showing a similar characteristic pattern for each patient with a rapid increase during the initial period, followed by gradual stabilization. Post-hoc comparisons revealed significant differences between appointments up to the fourth fitting session for M/C levels and up to the third fitting session for T-levels. There were no significant differences after these fitting sessions. These findings were consistent across the different electrode segments (apical, medial, upper basal, lower basal).

Conclusions. The mean M/C- and T-levels stabilized after approximately 3 months and 5 weeks, respectively, showing equivalent trajectories between the two brands.

Figure 1. Longitudinal changes in M-/C- and T-levels



Optimizing Amplitude Modulation Enhancement for Music Transmission Through Cochlear Implants

Luuk Schipper (Rijksuniversiteit Groningen)*; Nouri Khalass (Golden Hearing B.V.); Deniz Başkent (University of Groningen/University Medical Center Groningen, Department of Otorhinolaryngology); Eleanor E. Harding (Center for Language and Cognition, University of Groningen); Etienne Gaudrain (Lyon Neuroscience Research Center, CNRS UMR5292, Inserm U1028, Université Lyon 1)

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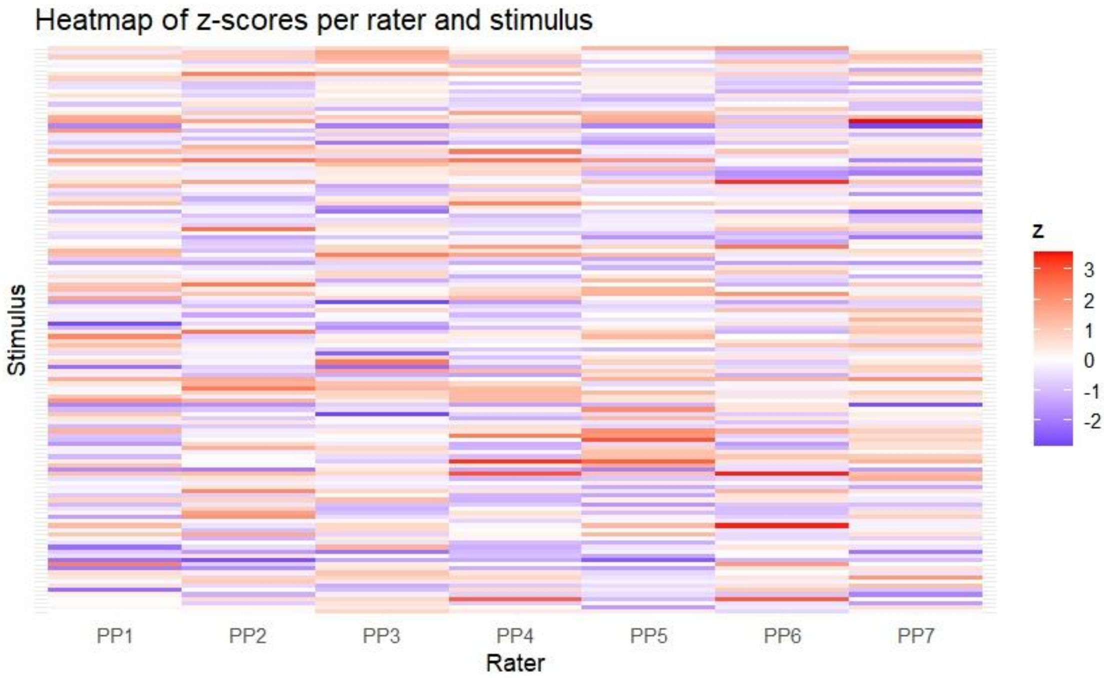
Poster

Abstract

Background. Cochlear implant (CI) users experience degraded music quality. The company Golden Hearing has developed an algorithm, the AM-Booster, for hearing aid users aiming to enhance music appreciation. The algorithm amplifies the magnitude of amplitude modulation (AM) in the signal, which is controlled by multiple parameters. While the company reports the algorithm to improve music quality for hearing aid users, the optimal parameters for a cochlear implant are not yet known. The parameter setting combination that provides the best musical experience might be determined by characteristics of specific musical excerpts, such as genre or the amount of pre-existing modulations. We investigate optimal AM-Booster parameters for CI users, and the relationship to musical genre.

Methods. 150 musical excerpts of 10 s, representing 15 genres, were randomly selected from the Free Music Archive database. In initial analysis steps, the level of the excerpts was adjusted to match a desirable listening level, based on the judgments of 7 raters, and the stimuli set was reduced to exclude items with prolonged silence, broadband noise, or speech without background music. The next planned steps include presenting the corpus to a CI-processor, with the AM-Booster algorithm applied with different parameter settings. The electrical stimulation produced by the implant will then be examined to evaluate the effect of the Booster. From previous work, two parameters were seen to influence the AM most. Therefore, the sampling of these parameters will be the most finely grained, with only sparse adjustments to the others.

Results. The excerpts will be clustered based on their modulation spectra and we will inspect the relationship to genre. Finally, we will use this to determine usable parameter settings. In a separate planned follow-up study, these parameter settings will undergo quality ratings by cochlear implant users to see whether the AM Booster can improve music listening experience.


Heatmap of z-scores per rater and stimulus
Stimulus
PP1
PP2
PP3
PP4
PP5
PP6
PP7
Rater
z
3
2
1
0
-1
-2

Towards modelling the impact of spread of excitation on classical psychophysical paradigms with cochlear implant users

Rutendo Chatiza (Cambridge Hearing Group, MRC Cognition and Brain Sciences Unit, University of Cambridge)*; Lidea Shahidi (Cambridge Hearing Group, MRC Cognition and Brain Sciences Unit, University of Cambridge); Robert P. Carlyon (Cambridge Hearing Group, MRC Cognition and Brain Sciences Unit, University of Cambridge); Francois Guérit (Cambridge Hearing Group, MRC Cognition and Brain Sciences Unit, University of Cambridge)

rcc62@cam.ac.uk

Poster

Abstract

Background. Cochlear implant (CI) users typically understand speech in quiet conditions, but in noisy conditions even the most successful struggle. One contributing factor is channel interactions, whereby the electrical fields generated by multiple CI stimulating electrodes overlap, leading to unintended co-stimulation of auditory nerve fibres. Auditory-nerve computational models could help us design and evaluate stimulation methods that reduce channel interactions, allowing rapid testing of varied stimuli. However, these models are inspired from animal data and are rarely validated against CI psychophysical literature. Here, we validate a computational model by comparing it to several published CI psychophysical datasets.

Methods. We select papers from the CI psychophysical literature with within-participant manipulations (e.g. the effect of temporal and spatial separation between two pulse trains on loudness summation) that investigated interactions both within and across channels. We then simulate them via the model by Brochier et al. (2022, IEEE TBME 69(11)) and test different metrics to compare the data against the modelled neural activation patterns. Finally, we investigate the impact of changing the model selectivity and time constants (refractoriness, central integration) on the fit between modelled neural activity and psychophysical data.

Results. Ongoing results show modelled outputs that are overall in range of the data found in the literature, as well as expected and unexpected differences between the modelled and published within-participant effects. This suggests necessary improvements in the model parameters and/or improvements in the metrics used to map modelled neural activity to psychophysical data.

Conclusions. By improving the comparison between the output of CI computational models and known within-participant effects, computational models can improve our understanding of the complex impact of channel interactions on speech perception with CIs.

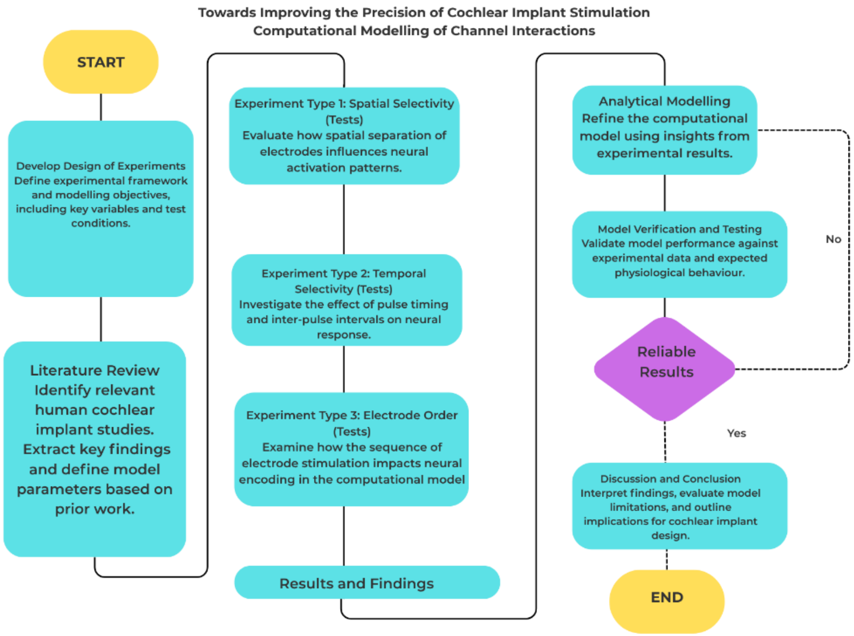


Figure 1: Workflow for improving the precision of cochlear implant stimulation using computational modelling of channel interactions. The study begins with the design of experiments, informed by a targeted literature review and first principles understanding of auditory nerve stimulation. Various experiments are conducted to investigate spatial selectivity, temporal selectivity, and electrode stimulation order, with results analysed after each stage. These findings are used to iteratively refine the model through analytical modelling, followed by verification and testing to ensure reliability. The process culminates in a validated model, supporting the interpretation of results and final conclusions.

Session 5.B. Hearing-Aid Innovation and Performance Assessment

Algorithms, real-ear analysis, and adaptive processing in modern devices.

Chaired by Dr. Raul Sanchez Lopez.

A Large-Scale Database and Predictive Model of Listener-Rated Ease of Speech Understanding in Commercial Hearing Aids

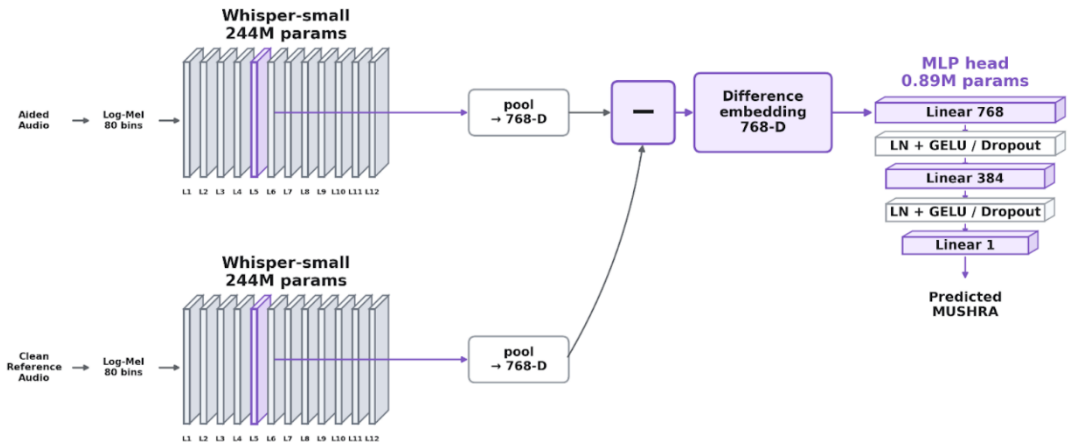
Andrew Sabin (HearAdvisor LLC)*; Steve Taddei (HearAdvisor LLC); Abram Bailey (HearAdvisor LLC)

andy@hearadvisor.com

Featured talk

Abstract

Predicting real-world hearing-aid benefit for speech understanding remains a central challenge in audiology. Most existing metrics focus on objective speech intelligibility, often with simulated devices and controlled lab conditions, which may not fully capture whether real users experience improvement in everyday listening. Here we present a large-scale perceptual database and predictive model of listener-rated ease of speech understanding, a measure related to perceived hearing-aid benefit. Ratings were collected from real hearing aid consumers listening to recordings from commercial hearing aids, each captured in the same set of spatially-realistic acoustic scenes. We deployed a blind listening test on our consumer-facing website, HearAdvisor.com, where participants rated how easy it was to understand speech on a five-point scale. The test was inspired by the MUSHRA paradigm, with high- and low-quality anchor signals on each trial (alongside four commercial devices). Since launch, we have collected over 120,000 ratings spanning more than 11,000 audio files from over 80 commercial hearing aid devices, each evaluated in 72 ambisonic scenes using recordings made on an acoustic manikin. To our knowledge, this is one of the largest perceptual datasets assembled for human evaluation of commercial hearing aids. We used this database to train a deep learning model that predicts listener-rated ease of speech understanding from audio. The best-performing architecture computed a difference embedding (aided minus clean-speech reference) from a pretrained automatic speech recognition model, passed through a small MLP trained to predict mean listener ratings. On held-out devices, the model generalized well ($r > 0.92$) and substantially outperformed a widely used objective intelligibility metric. These findings introduce a scalable approach to hearing-aid evaluation grounded in whether users perceive a device as making speech easier to understand.


Whisper-small
244M params
Aided
Audio
Log-Mel
80 bins
L1 L2 L3 L4 L5 L6 L7 L8 L9 L10 L11 L12
pool
→ 768-D
Difference
embedding
768-D
MLP head
0.89M params
Linear 768
LN + GELU / Dropout
Linear 384
LN + GELU / Dropout
Linear 1
Predicted
MUSHRA
Whisper-small
244M params
Clean
Reference
Audio
Log-Mel
80 bins
L1 L2 L3 L4 L5 L6 L7 L8 L9 L10 L11 L12
pool
→ 768-D

Performance Assessment for DNN-based Speech Enhancement Algorithms

Stefan Raufer (Sonova AG)*

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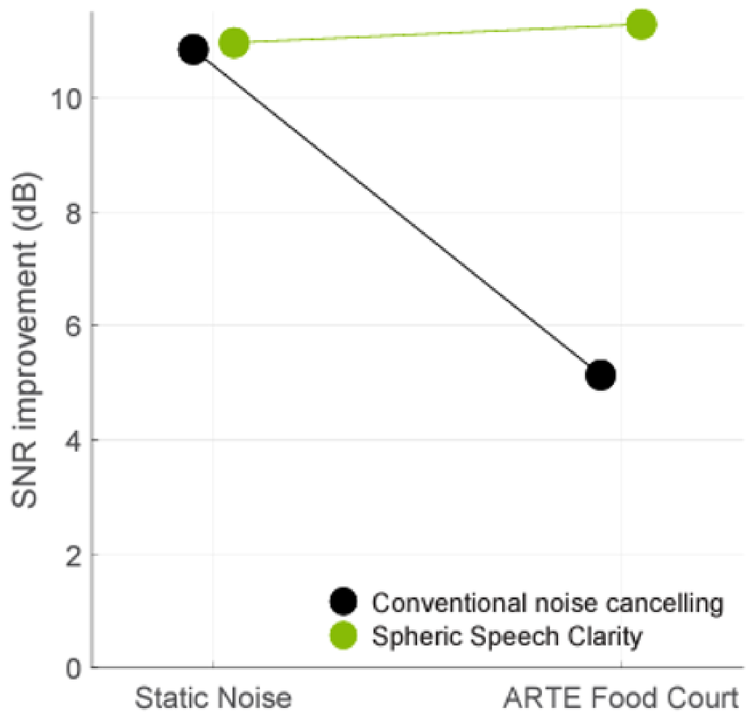
Podium

Abstract

Speech understanding in noise remains the primary challenge for individuals with hearing loss and a central target for hearing aid technology. Over the past forty years, advances in directional microphones and statistical noise cancellation have delivered substantial improvements in signal to noise ratio (SNR). More recently, deep neural network (DNN)–based speech enhancement has enabled a fundamental shift in algorithmic capabilities, offering robust performance even in complex and dynamic acoustic environments.

Despite these technological advances, outcome measures used to assess hearing aid performance have changed little. Metrics such as speech intelligibility, SNR improvement, and user preference remain dominant. In addition, test setups often rely on static noise sources, limited spatial complexity, and favorable SNRs. While well suited to demonstrate the benefits of classical signal processing approaches, simplified test setups tend to overestimate algorithmic benefit and fail to reflect the benefit in real world listening.

In this talk, I will demonstrate how the estimated SNR improvement depends on the spatial complexity, noise type, and input SNR of the background noise. Under simplified conditions, classical noise cancellation approaches can perform comparably to or better than DNN-based speech enhancement. However, in more complex environments, DNN-based approaches provide superior benefit. The findings demonstrate that simplified test paradigms obscure the conditions under which DNN-based algorithms outperform classical methods. To keep pace with technological progress, the field is moving toward more ecologically valid test methods, advanced modeling approaches, and outcome measures that extend beyond SNR and speech intelligibility and toward quality of life relevant measures.


SNR improvement (dB)
10
8
6
4
2
0
Conventional noise cancelling
Spheric Speech Clarity
Static Noise
ARTE Food Court

Technical Evaluation of Scene-Aware Dynamic Compression for Hearing Aids Using a Database of Natural Conversations

Moritz Bender (CvO Universität Oldenburg)*; Giso Grimm (CvO Universität Oldenburg); Volker Hohmann (CvO Universität Oldenburg)

moritz.bender@uni-oldenburg.de

Podium

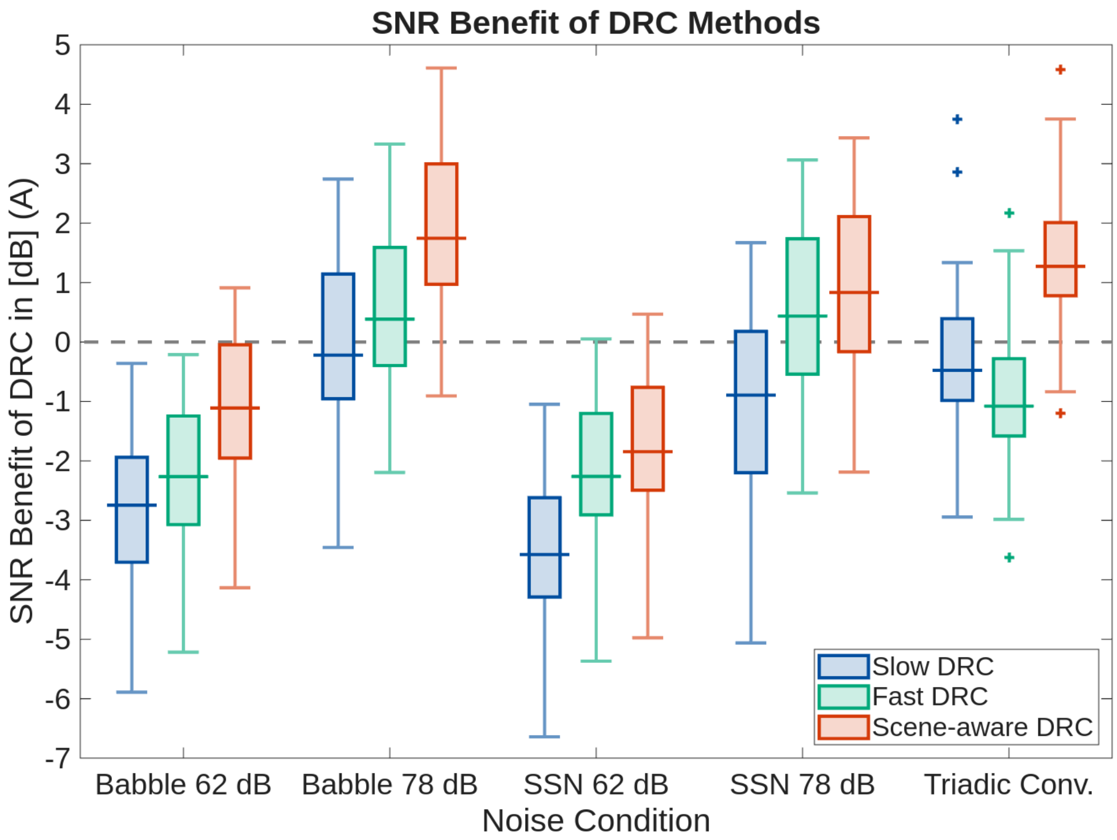
Abstract

Background. Hearing aids aim to improve the daily experience of hearing impaired (HI) listeners with a focus on speech reception. They compensate for effects of hearing loss through amplification, dynamic range compression (DRC) and noise reduction. However, their performance in challenging environments with multiple talkers and background noise still has large potential for improvement. To this end, scene-aware DRC and neural DRC have been proposed. They aim at providing different types of DRC for segregated foreground speech and background noise components in a scene to enhance the perception of foreground speech. Nevertheless, they are limited by the performance of source separation between foreground and background signals. Also it remains unclear, which combination of foreground and background processing is most beneficial for HI listeners in real-life communication scenarios, in which speech is a common noise component.

Methods. This study uses a database of free triadic conversations in various noise conditions. It contains movement data and separately available speech and background noise signals. This a priori knowledge enables a simulation of the conversations in a virtual acoustic environment. The resulting, separated signals at each receiver are then processed with implementations of scene-aware DRC (individual processing of foreground and background signals) and conventional DRC.

Results. The SNR benefit of processed signals is analyzed in five noise conditions: four of them are diffuse background noise (62 dB and 78 dB SPL babble noise and speech shaped noise) and one is speech dominated noise (a competing triadic conversation). Scene-aware DRC shows the highest SNR benefit across all conditions.

Conclusions. The higher SNR benefit in diffuse and speech dominated background noise indicates that performance benefits of scene-aware DRC hold up in complex communication scenarios. This result will be investigated with perceptual tests in future work.


SNR Benefit of DRC Methods
SNR Benefit of DRC in [dB] (A)
5
4
3
2
1
0
-1
-2
-3
-4
-5
-6
-7
Babble 62 dB
Babble 78 dB
SSN 62 dB
SSN 78 dB
Triadic Conv.
Noise Condition
Slow DRC
Fast DRC
Scene-aware DRC

Individualized Amplitude Compression for Improved Listening Outcomes at Moderate to High Sound Levels

Lukas Jürgensen (University of Southern Denmark (SDU))*; Michal Fereczkowski (University of Southern Denmark (SDU)); Tobias Neher (University of Southern Denmark (SDU))

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Podium

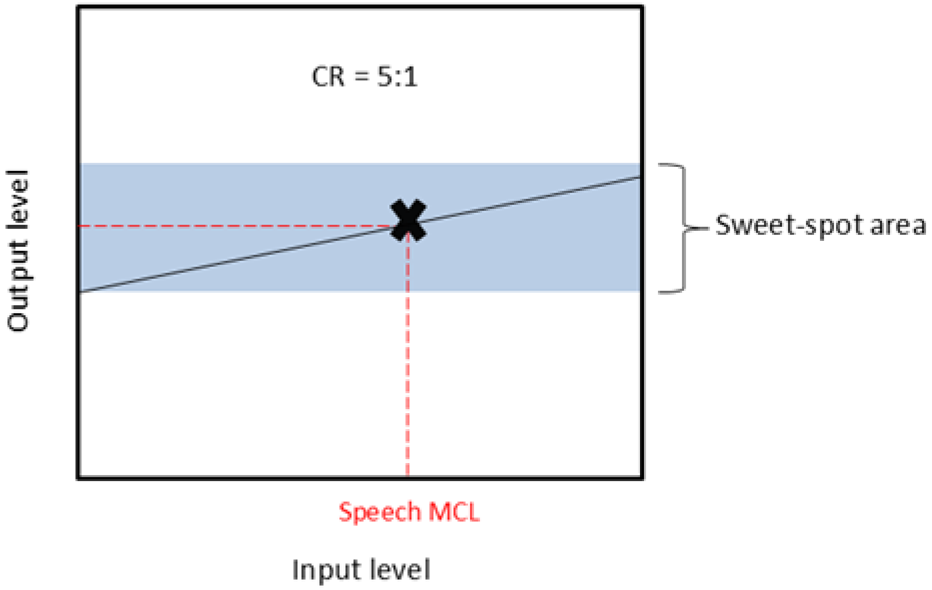
Abstract

Background. Hearing aid gain prescriptions are typically derived solely from the audiogram, thereby neglecting suprathreshold hearing abilities. This limitation may contribute to suboptimal aided outcomes, such as decreasing speech intelligibility (SI) at moderate to high sound levels (i.e., “rollover”). To address this issue, we previously introduced a fitting strategy called ‘sweet-spot compression’ (SSC). SSC aims to prevent rollover by placing speech in that area of an individual's performance-intensity function where both SI and listening comfort are high. To achieve this, SSC combines high compression ratios with long time constants to provide quasi-linear amplification around the individual most comfortable speech level.

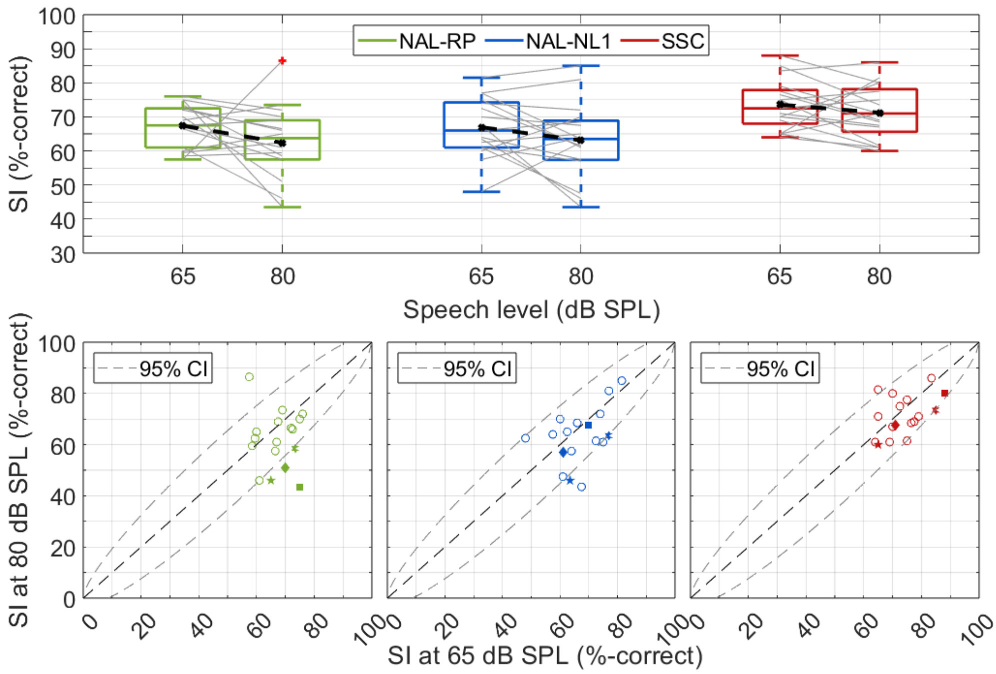
Methods. Using a hearing aid simulator, headphone presentation, and participants with clear rollover, we previously found that SSC outperformed a NAL-NL1-based reference condition. Here, we extended this work by implementing SSC on a wearable research hearing aid with optimized gain prescription, and by evaluating it with 17 randomly chosen hearing-impaired listeners. We collected SI scores in noise at 65- and 80-dB-SPL speech level, and pairwise preference judgments at multiple levels in quiet and in noise.

Results. Relative to the reference condition, SSC gave better SI at both presentation levels and was clearly preferred in quiet and in noise. At the group level, rollover occurred with neither SSC nor the reference condition. At the individual level, we found fewer cases of rollover with SSC than with the reference condition.

Conclusions. SSC shows promise with respect to improving listening outcomes at moderate to high sound levels.



CR = 5:1
Sweet-spot area
Output level
Speech MCL
Input level



NAL-RP
NAL-NL1
SSC
SI (%-correct)
Speech level (dB SPL)
95% CI
SI at 80 dB SPL (%-correct)
SI at 65 dB SPL (%-correct)

Integrating Neural Network Denoisers in Hearing Aids Under Real-Time Constraints

Clara Yaiche (AAL)*; Boltzmann Li (Aizip)

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Podium

Abstract

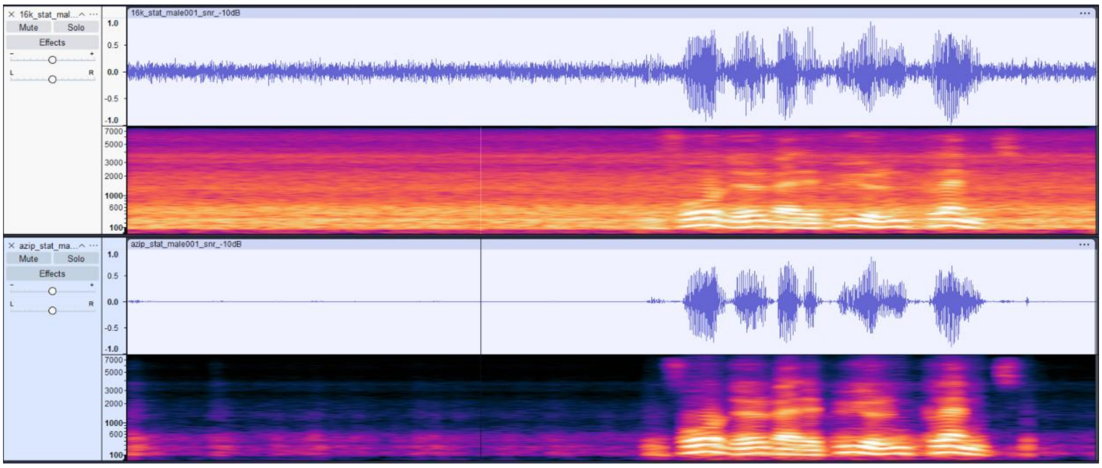
Hearing aids primarily restore audibility through amplification calibrated to the patient's audiogram. Amplification alone is not sufficient to improve speech intelligibility, which not only depends on signal-to-noise ratio (SNR). To improve intelligibility, most hearing aid software integrates denoisers to limit the noise before applying the patient's personalised gain.

Recent advances in AI have led to breakthroughs in denoising performance. However, most AI denoisers do not match real-time and computation constraints of edge audio devices. This is even more the case with hearing aids, where the total latency must remain below the echo perception threshold. Beyond latency, speech quality is also a requirement. Sadly a lot of AI-based denoisers only focus on noise removal and have negative impacts on speech signal quality. This remains inconceivable for hearing aid users, where speech understanding is so crucial. Furthermore, most AI denoiser entirely remove the noise which is information needed by the hearing impaired.

Therefore, integrating a neural network-based denoiser is both a technical challenge of edge AI processing and a speech enhancement step forward for the hearing impaired. However, it must be done with extreme precaution to conserve speech quality and intelligibility.

In this presentation we will show how edge-AI neural network can achieved such a breakthrough. We will detail how Aizip was able to integrate this neural network in Absolute Audio Labs' hearing aid software chain. The submission will explain the requirements of such an integration for real-time speech processing, what challenges are, and how to evaluate performances based on different metrics and perceptual testing.

This work demonstrates how a state-of-the-art AI denoiser can improve speech intelligibility in hearing aids and how it can benefit hearing aid users.


16k_stat_male001_snr_-10dB
azip_stat_male001_snr_-10dB
Mute
Solo
Effects

Participant-in-the-Loop Research on Speech Enhancement for Multi-Talker Scenarios in Hearing Aids

Melika Kianian (Auditory Signal Processing & Hearing Devices and Cluster of Excellence "Hearing4all.connects", Department of Medical Physics and Acoustics, University of Oldenburg, Oldenburg, Germany)*; Hendrik Kayser (Auditory Signal Processing & Hearing Devices and Cluster of Excellence "Hearing4all.connects", Department of Medical Physics and Acoustics, University of Oldenburg, Oldenburg, Germany); Volker Hohmann (Auditory Signal Processing & Hearing Devices and Cluster of Excellence "Hearing4all.connects", Department of Medical Physics and Acoustics, University of Oldenburg, Oldenburg, Germany)

Melika.kianian1@uni-oldenburg.de

Podium

Abstract

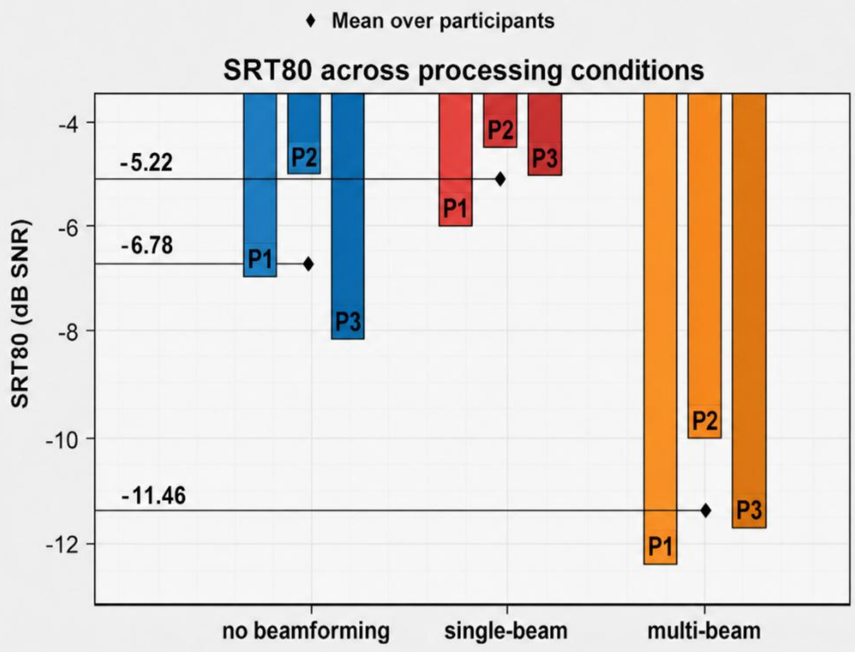
Background. A critical aspect of hearing aids is speech enhancement in challenging multi-talker environments with background noise. To this end, our previous studies simulated single- and multi-beam minimum variance distortionless response (MVDR) beamforming steered by direction-of-arrival (DOA) information. Using oracle knowledge with imposed estimation errors, our findings support the use of multi-beam MVDR beamforming in multi-talker scenarios.

Methods. The present work advances this approach to participant-in-the-loop operation. Real-time DOA estimation was implemented, replacing prior oracle DOA information, and beamforming was applied during a speech-reception task with turn-taking talkers. openMHA was used for hearing-aid signal processing, realizing three processing conditions: no beamformer, single-beam MVDR, and multi-beam MVDR. The TASCAR software was used to render an acoustic scene with two target talkers at ±30° in diffuse cafeteria noise. Pilot measurements were conducted with three participants using the German Matrix Sentence Test, implemented as two interleaved adaptive sentence-test tracks, one assigned to each target talker. Audio signals, DOA estimates, and behavioral responses were recorded. Performance was evaluated using Speech Reception Threshold at 80% intelligibility (SRT80) and offline signal-level analyses, including SNR improvement.

Results. The real-time DOA-guided MVDR framework was successfully implemented and tested. Pilot results showed the best behavioral outcome for multi-beam MVDR compared with single-beam MVDR and no beamforming. The logged data allow post-analysis of DOA steering behavior and subject performance.

Conclusions. This work demonstrates the suitability of the proposed framework for participant-in-the-loop evaluation of signal enhancement for hearing aids. The results

support multi-beam MVDR as a promising approach for robust speech enhancement in realistic multi-talker listening conditions.



SRT80 comparison across processing conditions. Speech reception thresholds were measured using the German Matrix Sentence Test with three pilot participants. Results are shown for no beamforming, single-beam MVDR, and multi-beam MVDR. Lower SRT80 values indicate better speech intelligibility.

Do Premium Hearing Aids Offer Better Outcomes Than Basic Models? A Systematic Review and Meta-Analysis

Nausheen Dawood (University of Pretoria)*; De Wet Swanepoel (University of Pretoria); Vinaya Manchaiah (University of Pretoria); Faheema Mahomed Asmail (University of Pretoria); Ilze Oosthuizen (University of Pretoria)

nausheendawood@gmail.com

Poster

Abstract

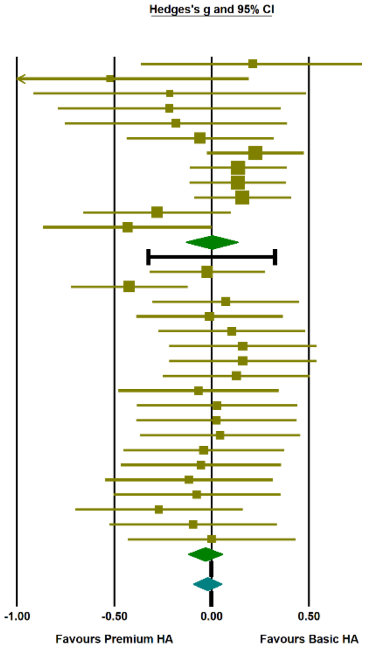
Background. Hearing aid technology ranges from basic to premium devices, with premium models incorporating advanced signal processing features intended to improve speech understanding and sound quality. Whether these higher technology levels lead to clinically meaningful benefits remains uncertain, particularly in the context of accessibility and affordability of hearing care.

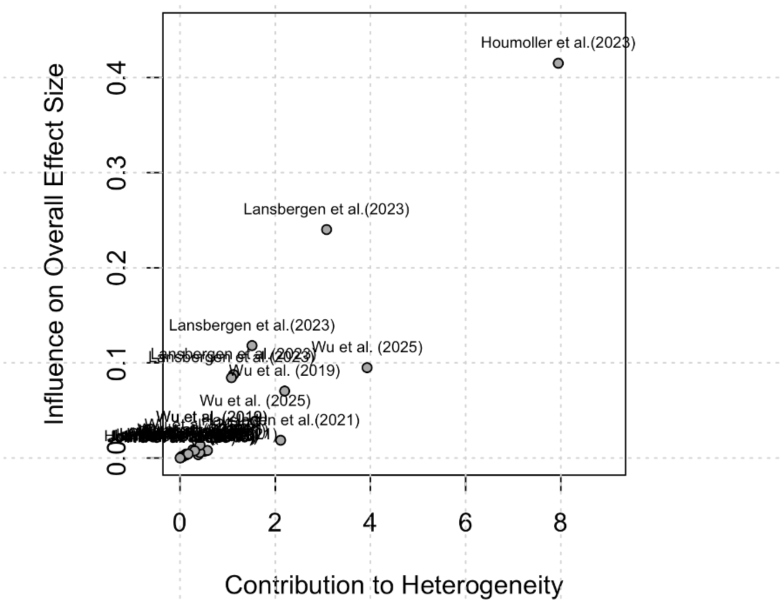
Methods. We conducted a systematic review and meta-analysis following PRISMA guidelines, with study selection guided by the PICOST framework. A comprehensive search of four electronic databases identified studies comparing outcomes associated with premium and basic hearing aids. Outcomes included self-reported measures, behavioural performance measures, and real-ear measurements. Included studies comprised crossover and parallel randomized controlled trials, cross-sectional studies, quasi-experimental designs, and studies with unspecified designs.

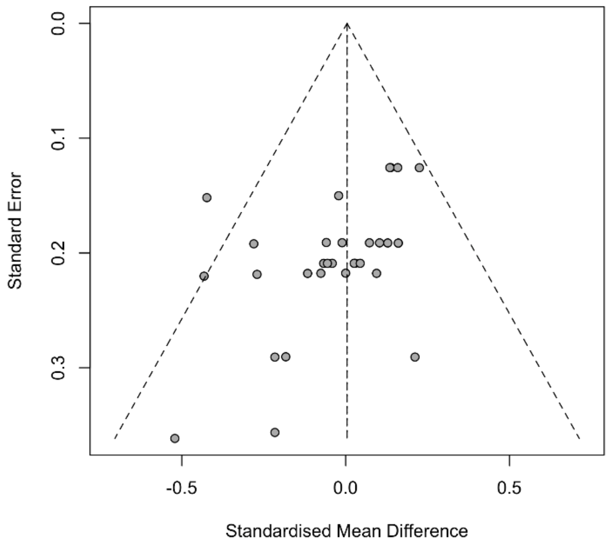
Results. Fifteen studies met the inclusion criteria, with nine studies contributing data from 7 952 participants to the meta-analysis. Across pooled self-reported and behavioural outcomes, no significant overall advantage was observed for premium hearing aids compared with basic models. While some studies reported modest benefits for premium devices in speech understanding and listening effort under controlled conditions, others showed minimal or no differences. Evidence from real-ear measurements was limited, with one study favouring premium devices. Overall, findings support the non-inferiority of basic hearing aids, with outcomes varying by listening context and individual needs.

Conclusions. These findings challenge assumptions that higher hearing aid technology levels necessarily produce superior outcomes and emphasize the importance of individualized clinical decision-making. Future research should standardize reporting of hearing aid technology classifications and further investigate the real-world impact of advanced features.

Comparison	Study name	Outcome	Hedges's g	Standard error	Variance	Lower limit	Upper limit	Z-Value	p-Value
			Statistics for each study						
Behavioural	Saleh et al.(2021)	Consonant identification	0.212	0.291	0.084	-0.358	0.781	0.728	0.467
Behavioural	Hausladen et al.(2021)	HINT	-0.521	0.362	0.131	-1.230	0.187	-1.442	0.149
Behavioural	Hausladen et al.(2021)	QuickSIN	-0.216	0.356	0.127	-0.914	0.483	-0.606	0.544
Behavioural	Plyler et al.(2021)	HINT	-0.216	0.291	0.085	-0.786	0.354	-0.743	0.457
Behavioural	Plyler et al.(2021)	QuickSIN	-0.183	0.290	0.084	-0.752	0.386	-0.629	0.529
Behavioural	Cox et al.(2014)	AFAAF	-0.059	0.191	0.036	-0.433	0.315	-0.309	0.757
Behavioural	Lansbergen et al.(2023)	AVAB-Detection	0.225	0.126	0.016	-0.022	0.471	1.789	0.074
Behavioural	Lansbergen et al.(2023)	AVAB-Localization	0.137	0.126	0.016	-0.109	0.384	1.094	0.274
Behavioural	Lansbergen et al.(2023)	AVAB-Speech in-quiet	0.135	0.126	0.016	-0.111	0.381	1.074	0.283
Behavioural	Lansbergen et al.(2023)	AVAB-Speech-in-noise	0.159	0.126	0.016	-0.087	0.405	1.264	0.206
Behavioural	Wu et al. (2019)	HINT	-0.280	0.192	0.037	-0.657	0.096	-1.460	0.144
Behavioural	Wu et al. (2025)	CST	-0.432	0.220	0.048	-0.864	-0.001	-1.963	0.050
	Pooled		0.002	0.068	0.005	-0.131	0.134	0.022	0.982
	Prediction Interval		0.002			-0.323	0.326		
Self-reported	Houmoller et al.(2023)	IOI-HA	-0.021	0.150	0.023	-0.316	0.273	-0.142	0.887
Self-reported	Houmoller et al.(2023)	SSQ-12	-0.424	0.152	0.023	-0.721	-0.126	-2.790	0.005
Self-reported	Wu et al. (2019)	APHAB	0.073	0.191	0.037	-0.302	0.447	0.380	0.704
Self-reported	Wu et al. (2019)	EMA-global	-0.011	0.191	0.037	-0.385	0.364	-0.057	0.955
Self-reported	Wu et al. (2019)	SSQ -Spatial	0.103	0.191	0.037	-0.271	0.478	0.541	0.588
Self-reported	Wu et al. (2019)	SSQ -Speech	0.161	0.191	0.037	-0.214	0.536	0.840	0.401
Self-reported	Wu et al. (2019)	SSQ-Listening Effort	0.160	0.191	0.037	-0.215	0.536	0.838	0.402
Self-reported	Wu et al. (2019)	SSQ-Quality	0.128	0.191	0.037	-0.247	0.503	0.671	0.502
Self-reported	Johnson et al. (2016)	APHAB-HA 1	-0.067	0.209	0.044	-0.477	0.343	-0.322	0.748
Self-reported	Johnson et al. (2016)	APHAB-HA 2	0.029	0.209	0.044	-0.381	0.438	0.137	0.891
Self-reported	Johnson et al. (2016)	LE-SSQ, DOSO-HA1	0.026	0.209	0.044	-0.384	0.436	0.124	0.902
Self-reported	Johnson et al. (2016)	LE-SSQ, DOSO-HA2	0.044	0.209	0.044	-0.365	0.454	0.212	0.832
Self-reported	Johnson et al. (2016)	Speech Composite Score-HA 1	-0.041	0.209	0.044	-0.451	0.369	-0.196	0.845
Self-reported	Johnson et al. (2016)	Speech Composite Score-HA 2	-0.056	0.209	0.044	-0.465	0.354	-0.266	0.790
Self-reported	Wu et al. (2025)	GHABP (Retro)	-0.116	0.218	0.047	-0.543	0.311	-0.533	0.594
Self-reported	Wu et al. (2025)	GPHAB (EMA)	-0.076	0.218	0.047	-0.502	0.351	-0.348	0.728
Self-reported	Wu et al. (2025)	HHIE/A	-0.270	0.219	0.048	-0.699	0.158	-1.237	0.216
Self-reported	Wu et al. (2025)	PHAB	-0.094	0.218	0.047	-0.521	0.332	-0.433	0.665
Self-reported	Wu et al. (2025)	SADL	0.000	0.218	0.047	-0.427	0.427	0.000	1.000
	Pooled		-0.032	0.045	0.002	-0.120	0.056	-0.706	0.480
	Prediction Interval								
	Pooled		-0.022	0.037	0.001	-0.095	0.052	-0.575	0.565
	Prediction Interval								







Adult Hearing Aid Users’ Perspectives on Outcome Expectations and Perceived Value for Money: A Qualitative Study Using the Health Belief Model

Nausheen Dawood (University of Pretoria)*; De Wet Swanepoel (University of Pretoria); Faheema Mahomed Asmail (University of Pretoria); Ilze Oosthuizen (University of Pretoria); Vinaya Manchaiah (University of Pretoria)

nausheendawood@gmail.com

Poster

Abstract

Background. Hearing aids are the primary intervention for hearing loss, yet disparities in use and user experience remain. Limited research has explored how outcome expectations and perceived value for money influence these experiences, particularly within the Health Belief Model (HBM).

Methods. This qualitative descriptive study examined adult hearing aid users’ expectations and perceptions of value for money using the HBM as a guiding framework. Semi-structured interviews were conducted with 35 adult hearing aid users, both new and experienced, with a mean age of 67.2 years. Participants were recruited via the Hearing Tracker website, and data were analysed using content analysis.

Results. Two domains emerged: Expectations and Perceived Value for Money. Fulfilled expectations and higher perceived value were associated with improved hearing, greater environmental awareness, device customisation, and user self-efficacy. Participants with realistic expectations reported confidence in managing their hearing aids and satisfaction with outcomes. In contrast, unfulfilled expectations and reduced use were linked to perceived barriers, including device limitations, difficulty hearing in background noise, physical fit challenges, and affordability concerns. While some participants viewed hearing aids as a worthwhile investment, others described financial strain related to high costs and limited insurance coverage.

Conclusions. Within the HBM framework, perceived benefits and self-efficacy supported positive hearing aid experiences, whereas performance, fit, and financial barriers undermined outcomes. Addressing expectation management, skills development, and affordability may improve sustained hearing aid use.

Table 2. Qualitative categories and sub-categories shaping hearing aid expectations as perceived outcomes within the HBM.

Category (n)	Sub-category (n)	HBM Constructs	Meaning Unit Examples (Participant ID, Age in Years, Gender)
Experiences contributing to fulfilled expectations (Sub domain)			
User Perspectives (39)	Self-perceived benefit (15)	Perceived Benefit	*"They improved my hearing. Definitely could hear better." (P26,64, M)*
	Overcoming stigma (11)	Perceived Barrier Self-Efficacy	*"I've just sort of accepted it more" (P27,68, F)*
	Awareness of hearing loss (6)	Perceived Severity	*"It was difficult for me to hear everybody." (P27,68, F)*
	Realistic expectations (4)	Perceived Benefit Self-Efficacy	*"I knew that it was not going to make my hearing what it used to be" (P32,81, F)*
	Perceived cost–benefit alignment informing expectations (3)	Perceived Benefit	*"There's a cost benefit" (P33,68, M)*
Device related facilitators (20)	Advanced technology (8)	Perceived Benefit	*"I was impressed with the technology" (P14,73, F)*
	Streaming and connectivity (5)	Self-Efficacy Perceived Benefit	*"I could have the Bluetooth on my phone and… I could hear the person I was talking to on the phone directly in my ear. "(P14,73, F)*
	App-based adjustments (4)	Self-Efficacy	*"I can control the volume. I can focus the hearing aid a little bit more on people that are talking to me... There's a wide range of things you can do because of the app" (P13,73, M)*
	Sound quality and Physical Fit (3)	Perceived Benefit	*"They're always very comfortable…, you don't even realize you have them on most of the time." (P25,71, M)*
Experiences contributing to unfulfilled expectations (Sub domain)			
Device related barriers (25)	Device limitations (12)	Perceived Barrier	*"There are still some limitations to what the hearing aid can do" (P12,60, F)*
	Background noise challenges (7)	Perceived Barrier	*"I always wished that those situations of hearing in noise was better than what it was" (P22,56, M)*
	Physical fit and handling (6)	Perceived Barrier Self-Efficacy	*"I find a little frustrating to use them" (P3,84, M)*

n= number of meaning units/mentions (multiple codes per participant were possible)

Table 3. Qualitative categories and sub-categories shaping hearing aid user perceptions on value for money within the HBM.

Category (n)	Subcategory (n)	HBM Constructs	Meaning Unit Examples (Participant ID, Age in Years, Gender)
Experiences contributing to perceived value for money (Sub domain)			
Value for Money (26)	Cost-benefit satisfaction (12)	Perceived Benefits	*"Yeah, I'd say that the money was worth it." (P15,46, M)*
	Advanced technology (9)	Perceived Benefits	*"Yeah, I'm willing to pay a little high if I… get top grade stuff… the features that are available on top of the line it's worth it to me" (P33,68, M)*
	HCP service delivery (5)	Perceived Benefits	*"a lot of it has to do with the fitting and all that kind of stuff" (P18,74, M)*
Perspectives on hearing aid impact (18)	Self-perceived benefit (12)	Perceived Benefits	*"Then, when you realize that you can hear things again that you'd forgotten about. And I mean it, it is the most amazing thing" (P8,63, F)*
	Importance of hearing (6)	Perceived Severity	*"I can't get along without it." (P24,82, M)*
Access to finance (8)	Additional Financial Resources (5)	Perceived Benefits (Reduced)	*"I'm also fortunate that I could afford hearing aids if I didn't." (P13,73, M)*
	Insurance coverage (3)	Perceived Barriers (Reduced)	*"I am still not paying much out of pocket" (P5,71, M)*
Experiences limiting perceived value for money (Sub domain)			
Financial Constraints (14)	Unaffordability (9)	Perceived Barriers	*"Hearing aids are incredibly expensive" (P19,64, M)*
	Financial barrier (2)	Perceived Barriers	*"I wish I didn't have to spend as much." (P12,60, F)*
Device-related constraints (3)	Technical challenges (3)	Perceived Barriers	*"Let's say all the problems I've been having with this particular brand... This set has been of such problem" (P31,82, F)*

Note: HCP= health care professional; n= number of meaning units/mentions (multiple codes per participant were possible)

Voice cues and vocal emotions perception in adult hearing aid users

Kateryna Skupovska (University of Groningen / University Medical Center Groningen)*; Laura Rachman (University of Groningen / University Medical Center Groningen); Sander Ubbink (University of Groningen / University Medical Center Groningen); Deniz Başkent (University of Groningen / University Medical Center Groningen)

k.skupovska@rug.nl

Poster

Abstract

Background. In listeners with normal hearing (NH), voice perception and vocal emotion recognition develop early in life, refine during adolescence, and mature in adulthood. However, for individuals with hearing loss, especially hearing aid (HA) users, the development of voice perception has been relatively understudied. While HAs amplify surrounding sounds and improve audibility, they must also compensate for the ear's mechanisms affected by hearing loss. As a result, they can compress the original acoustic information, thereby modulating the voice cues. Previous research has shown delayed and incomplete development of voice cues perception in children with HAs. However, it is not yet known how these skills evolve in adulthood, whether they continue developing in a delayed manner, plateau, or decline over time.

Methods. The study includes two age-matched adult groups: non-users (control group) and users of HAs (target group). Participants complete experimental tasks assessing voice cues discrimination, voice gender categorisation, vocal emotion recognition, and speech-in-speech perception. Additional data is collected to account for variability and potentially relevant factors, including unaided and aided hearing thresholds, questionnaires on HA use and demographics, and a cognitive screening test.

Results. We expect that HA use and hearing status will affect the sensitivity to voice cues, resulting in lower accuracy across all experimental tasks. Also, we expect to find individual variability among HA users, with performance being predicted by age, severity of hearing loss, HA experience, and cognitive abilities.

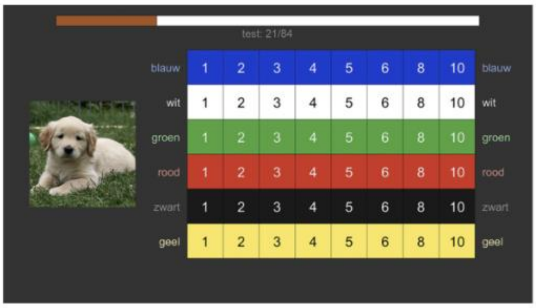
Conclusions. This study aims to improve understanding of how hearing loss and HA use affect voice and speech perception in adulthood. Addressing this knowledge gap has implications for theoretical models of auditory processing and can also impact clinical practice through improvements in counselling, rehabilitation, and the development of HA technology.

PICKA-XL (Perception of Indexical Cues in Kids and Adults) test battery

Voice cue discrimination

Voice gender categorisation

Vocal emotion recognition



Speech perception in competing speech

Session 6.A. From Lab to Clinic: Ensuring Impact of Computational Hearing Research

Translation, reproducibility, and open-science frameworks for clinical relevance.

Chaired by Dr. Ángel Ramos de Miguel.

PECAP Online: Translating the Panoramic ECAP Method to Clinic

Charlotte Garcia (University of Cambridge)*

charlotte.garcia@mrc-cbu.cam.ac.uk

Featured talk

Abstract

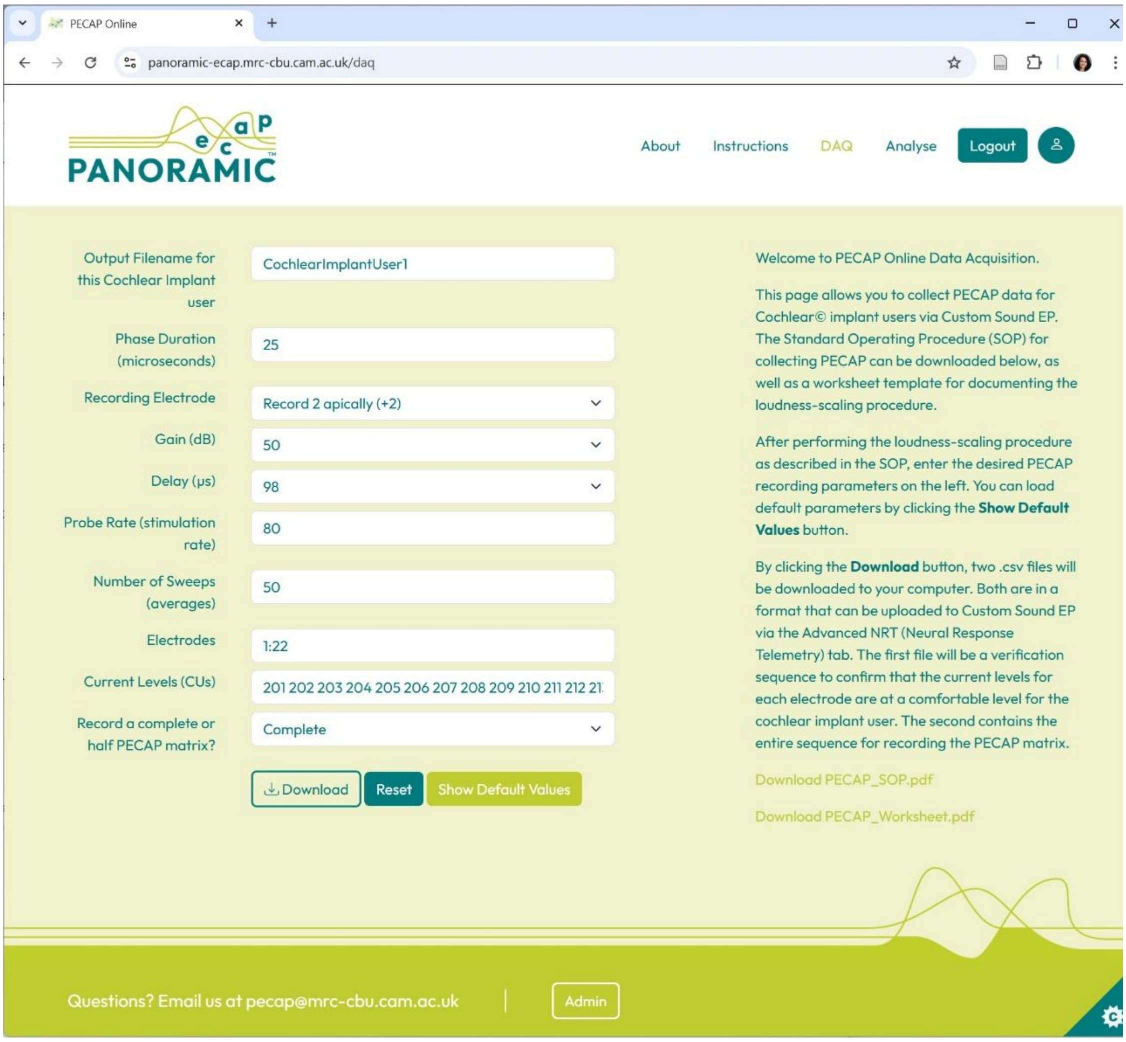
Background. The Panoramic ECAP Method (PECAP) uses Electrically Evoked Compound Action Potentials (ECAPs) to generate detailed patient-specific estimates of peripheral neural responsiveness and current spread for cochlear implant (CI) patients. It has detected localized areas of reduced neural responsiveness, manipulations to current spread, and differences in patterns of current spread between Cochlear © array types. This method can be leveraged for personalized programming of CI software, and may help individuals struggling to hear with their implants to reach their speech perception potential.

Method: Previously, PECAP was only available as a research tool and required specialized programming experience to operate. This project developed a website that allows users to collect and analyse PECAP data without the need for specialist, non-clinical software. The site has two primary functions: (1) users can enter recording parameters and download data collection files that can be imported into Cochlear ©'s "Custom Sound EP" clinical software to record the PECAP data, and (2) users can upload collected PECAP data and analyse it using the PECAP algorithm, extracting plots and data that describe the current-spread and neural-responsiveness estimates for their individual users.

Results. The online tool was launched in late 2024, and now has over 35 users (spanning researchers, industry representatives, and clinicians) from 10+ countries around the globe. Feedback is being collected in an ongoing manner to inform future updates, and expansion for use with MEDEL and Cochlear's NEXA devices.

Conclusion: Development of PECAP into an online platform not only enables external researchers to analyse their own PECAP data, but also allows clinicians to access the resource. We hope that this will increase the ability of clinicians to provide personalised hearing healthcare to cochlear-implant users.


PECAP Online
panoramic-ecap.mrc-cbu.cam.ac.uk
ecaP
PANORAMIC
About
Instructions
DAQ
Analyse
Logout
MAXIMISING HEARING POTENTIAL
THE PANORAMIC ECAP METHOD
Cochlear Implants are arguably the most successful neuro-prosthetic device in existence today, and can provide a sense of hearing to d/Deaf individuals by directly stimulating their auditory nerve. Each cochlear implant patient has a unique story and hearing status, and the ease with which they can communicate with their devices varies from person to person. This underlies a need for personalized hearing healthcare to maximize the hearing potential for individual users.
The Panoramic ECAP Method – or 'PECAP' – is a research tool that leverages objective measurements of neural responses to a cochlear implant and provides detailed estimates of variation in neural-activation patters along the length of the cochlea. In brief, it can provide a fingerprint of the way an individual cochlear implant patient's ear is communicating with their implant. It is our hope that PECAP can be used to optimize the software of cochlear implants for individual patients' unique patterns of hearing and enable them to better engage with the auditory world around them.
ABOUT THIS WEBSITE
Registering to this website will allow you to collect PECAP data from users of Cochlear devices. A Standard Operating Procedure document is available to provide you with instructions for use. The procedure will involve collecting a panorama of Electrically Evoked Compound Action Potentials (ECAPs) using the clinical software, Custom Sound EP. If you are interested in collecting PECAP data in users of Advanced Bionics or MED-EL devices, please contact us with your query, as additional research software will be required.
The website will also allow you to analyse PECAP data to estimate patient-specific patterns of neural responsiveness and current spread along the length of a cochlear implant electrode array.
Security is one of our highest priorities. We thank you for your patience with our approval process for registration to the PECAP website. You will receive an email when your registration request is approved with instructions for accessing the functional pages of the site.


PECAP Online
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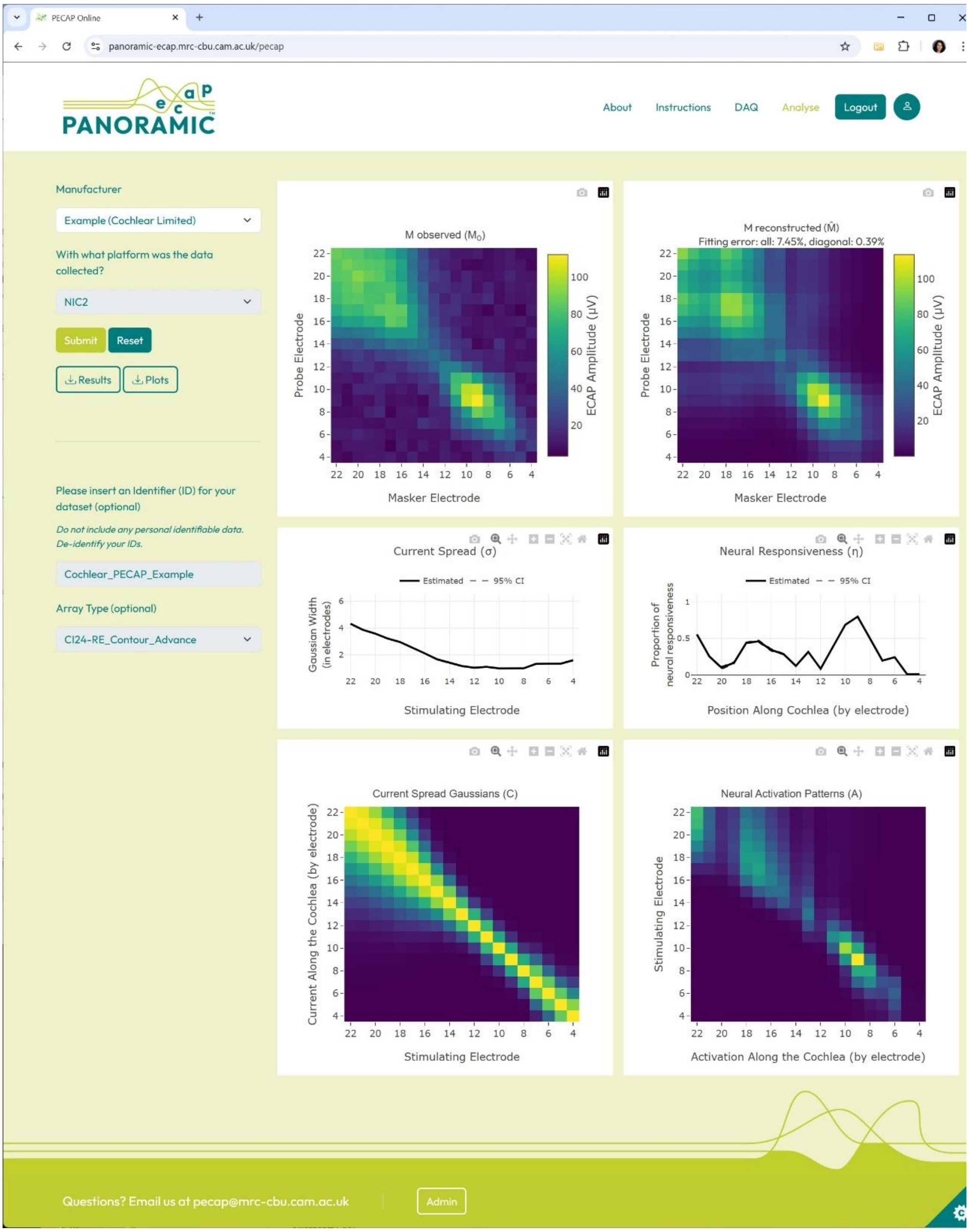
PANORAMIC
About
Instructions
DAQ
Analyse
Logout
Output Filename for this Cochlear Implant user
CochlearImplantUser1
Phase Duration (microseconds)
25
Recording Electrode
Record 2 apically (+2)
Gain (dB)
50
Delay (μs)
98
Probe Rate (stimulation rate)
80
Number of Sweeps (averages)
50
Electrodes
1:22
Current Levels (CUs)
201 202 203 204 205 206 207 208 209 210 211 212 21
Record a complete or half PECAP matrix?
Complete
Download
Reset
Show Default Values
Welcome to PECAP Online Data Acquisition.
This page allows you to collect PECAP data for Cochlear© implant users via Custom Sound EP. The Standard Operating Procedure (SOP) for collecting PECAP can be downloaded below, as well as a worksheet template for documenting the loudness-scaling procedure.
After performing the loudness-scaling procedure as described in the SOP, enter the desired PECAP recording parameters on the left. You can load default parameters by clicking the Show Default Values button.
By clicking the Download button, two .csv files will be downloaded to your computer. Both are in a format that can be uploaded to Custom Sound EP via the Advanced NRT (Neural Response Telemetry) tab. The first file will be a verification sequence to confirm that the current levels for each electrode are at a comfortable level for the cochlear implant user. The second contains the entire sequence for recording the PECAP matrix.
Download PECAP_SOP.pdf
Download PECAP_Worksheet.pdf
Questions? Email us at pecap@mrc-cbu.cam.ac.uk
Admin


PECAP Online
panoramic-ecap.mrc-cbu.cam.ac.uk/pecap
PANORAMIC
About
Instructions
DAQ
Analyse
Logout
Manufacturer
Example (Cochlear Limited)
With what platform was the data collected?
NIC2
Submit
Reset
Results
Plots
Please insert an Identifier (ID) for your dataset (optional)
Do not include any personal identifiable data. De-identify your IDs.
Cochlear_PECAP_Example
Array Type (optional)
CI24-RE_Contour_Advance
M observed (M_O)
M reconstructed ($\hat{M}$)
Fitting error: all: 7.45%, diagonal: 0.39%
Probe Electrode
Masker Electrode
ECAP Amplitude (µV)
Current Spread (σ)
Estimated
95% CI
Gaussian Width (in electrodes)
Stimulating Electrode
Neural Responsiveness (η)
Proportion of neural responsiveness
Position Along Cochlea (by electrode)
Current Spread Gaussians (C)
Current Along the Cochlea (by electrode)
Neural Activation Patterns (A)
Stimulating Electrode
Activation Along the Cochlea (by electrode)
Questions? Email us at pecap@mrc-cbu.cam.ac.uk
Admin

Reliability and Validity of the Audiometric Weber Test

Michael Finkelstein (Shamir Medical Center)*

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Podium

Abstract

Background. The Audiometric Weber Test was designed to verify BC hearing thresholds when they are suspected of being artificially low due to central masking. This study aimed to evaluate the intra-rater reliability and clinical validity of the Audiometric Weber Test. Additionally, the study investigated the impact of tympanic membrane perforation on test outcomes.

Methods. Seventy-eight patients with asymmetric conductive hearing loss were included in the data analysis. Each underwent a comprehensive behavioral hearing test followed by the Audiometric Weber Test by placing the bone conduction oscillator on the forehead. To assess intra-rater reliability, the test was repeated three times at each frequency (500, 1000, 2000, and 4000 Hz). Machine learning analysis was conducted to find more valid formula.

Results. Although the intra-rater reliability was found to be very good, its validity was only moderate (46.2%–58.7%). The results were similar when patients with greater asymmetry were analyzed. The Audiometric Weber Test found to be more accurate in patients with perforated tympanic membrane at 2000 Hz and 4000 Hz.

Conclusions. Although reliable, the current clinical validity of the Audiometric Weber Test is limited for identifying asymmetric conductive hearing loss. Furthermore, while machine learning models achieved higher accuracy, their complexity renders them impractical for routine clinical use, suggesting that objective audiological measures remain superior for verifying thresholds

Intra-rater Reliability as Measured by Kappa and Alpha Values of Absolute and Partial Consistency and Accuracy of the Formula

Test Frequency	500 Hz	1000 Hz	2000 Hz	4000 Hz
Number of participants	76	75	66	52
Krippendorff's Alpha	.96	.96	.95	.96
Fleiss' Kappa	.81	.77	.72	.78
Accuracy of the formula	46.2%	50%	50.7%	58.7%

Toward Ecologically Valid Hearing Assessment

Zahi Tubul (Kaplan Medical Center)*

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Podium

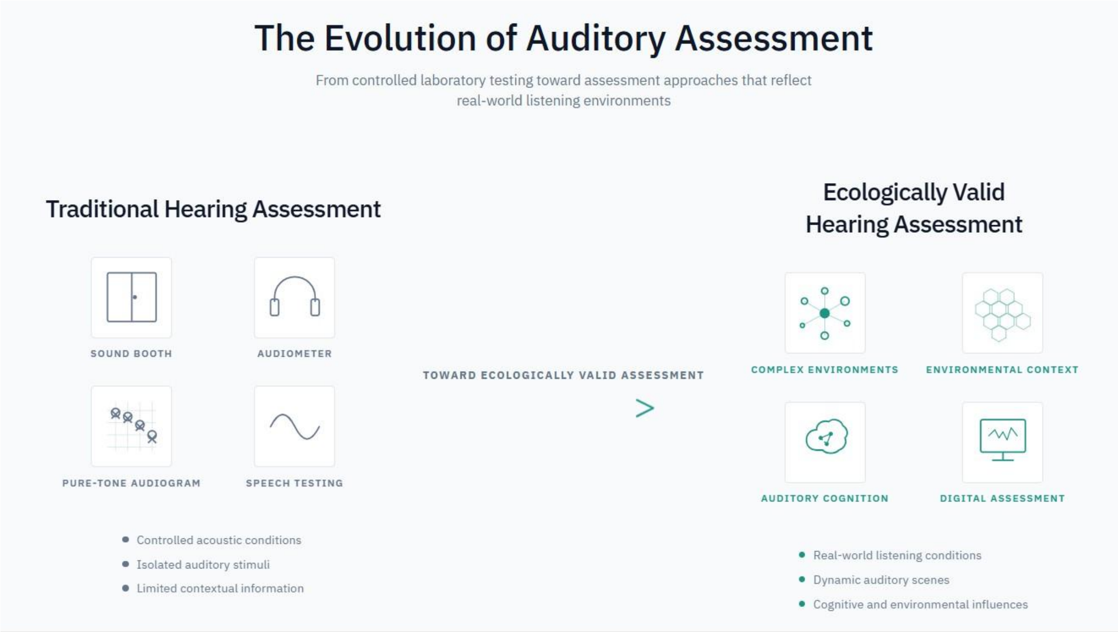
Abstract

Background. Traditional hearing assessment in audiology is typically conducted in highly controlled laboratory environments such as sound-treated booths. While these paradigms have enabled reliable measurement of auditory thresholds and speech perception, they may not fully capture the complexity of listening in everyday environments. Increasing attention in hearing science has therefore been directed toward improving the ecological validity of auditory assessment.

Methods. This work presents a conceptual perspective synthesizing emerging research directions aimed at bridging the gap between laboratory-based testing and real-world listening conditions. Drawing on studies in auditory scene analysis, cognitive hearing science, and digital hearing technologies, we examine how factors such as dynamic acoustic environments, cognitive load, and multisensory context influence auditory perception and listening performance.

Results. Across multiple research domains, a growing body of work highlights the limitations of traditional assessment paradigms when predicting real-world listening abilities. Recent approaches propose incorporating more complex acoustic scenes, context-aware testing paradigms, and digitally mediated assessment tools in order to better reflect natural listening conditions. These developments suggest a shift toward assessment models that consider hearing as an interaction between auditory processing, cognitive mechanisms, and environmental context.

Conclusions. Advancing toward ecologically valid hearing assessment may improve the relevance of diagnostic procedures and strengthen the connection between research findings and clinical practice. Conceptual frameworks that integrate auditory, cognitive, and environmental factors can help guide the development of next-generation assessment tools and support more realistic evaluation of hearing function.



The Evolution of Auditory Assessment
From controlled laboratory testing toward assessment approaches that reflect real-world listening environments
Traditional Hearing Assessment
SOUND BOOTH
AUDIOMETER
PURE-TONE AUDIOGRAM
SPEECH TESTING
Controlled acoustic conditions
Isolated auditory stimuli
Limited contextual information
TOWARD ECOLOGICALLY VALID ASSESSMENT
Ecologically Valid Hearing Assessment
COMPLEX ENVIRONMENTS
ENVIRONMENTAL CONTEXT
AUDITORY COGNITION
DIGITAL ASSESSMENT
Real-world listening conditions
Dynamic auditory scenes
Cognitive and environmental influences

Data Standards in Audiology: Data Models and Community Participation

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Podium

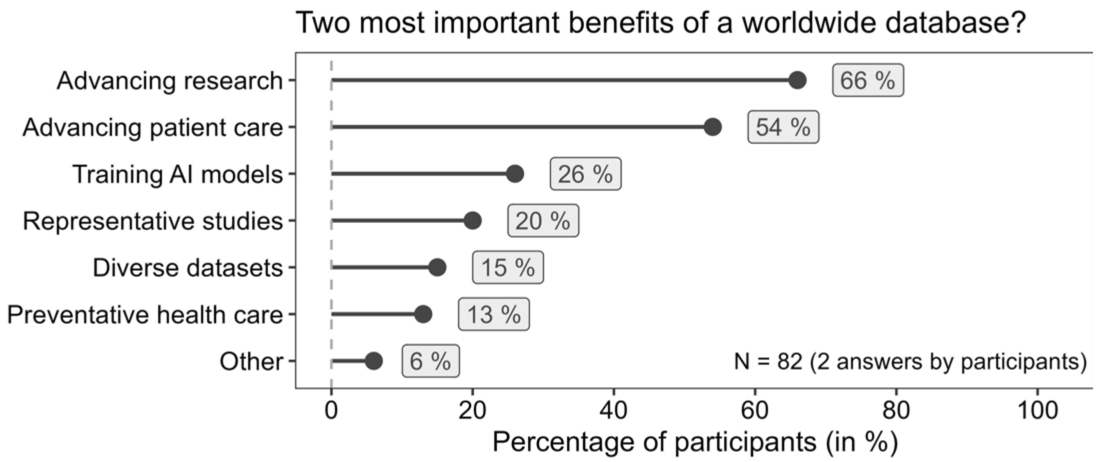
Abstract

Background. Data standards provide a means to harmonize data formats, to enhance structured documentation and, thus, combining and reusing data. In audiology, data from clinical routine, research studies or hearing device data logging yield a large potential to be exploited with artificial intelligence (Fig. 1), but differences in audiological tests and conditions, data formats or quality hinder data exchange and combination. Accessible, interoperable big audiological data would enable precise analysis and prediction, with the potential to advance research and patient care. An EFAS working group strives to establish data standards for audiology.

Methods. Three main data standardization approaches were identified by Vercammen et al. (2026), from which two are open and community-based. Open Electronic Health Record (openEHR) and the Observational Medical Outcomes Partnership (OMOP) are widely used in different medical fields, but they differ in purpose, concept, and properties. For both initiatives, developments in the field of audiology are ongoing.

Results. In openEHR, a data model for categorical loudness scaling is published, one for speech intelligibility is under community review, and modelling of the audiogram and additional audiological tests is ongoing. In OMOP, data models for audiogram data exist, and their utility for combining and analyzing data from different sources in the context of hearing assessment and hearing aid fitting is currently being explored in a proof-of-concept study.

Conclusions. Community-based data standardization in audiology has led to significant progress in both open data standards. As both approaches rely on extensive metadata to document audiological data, future convergence remains possible based on transformations between standards. To further pursue and strengthen the efforts, the audiological community is invited to participate by providing their expertise and different perspectives, e.g., by reviewing data models.



From Detection to Understanding: Speech-in-Noise Screening Identifies Functional Auditory Deficits with Electrophysiologic Correlates in a Clinical Subset

Jacqueline Scholl (Soundwrx)*

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Podium

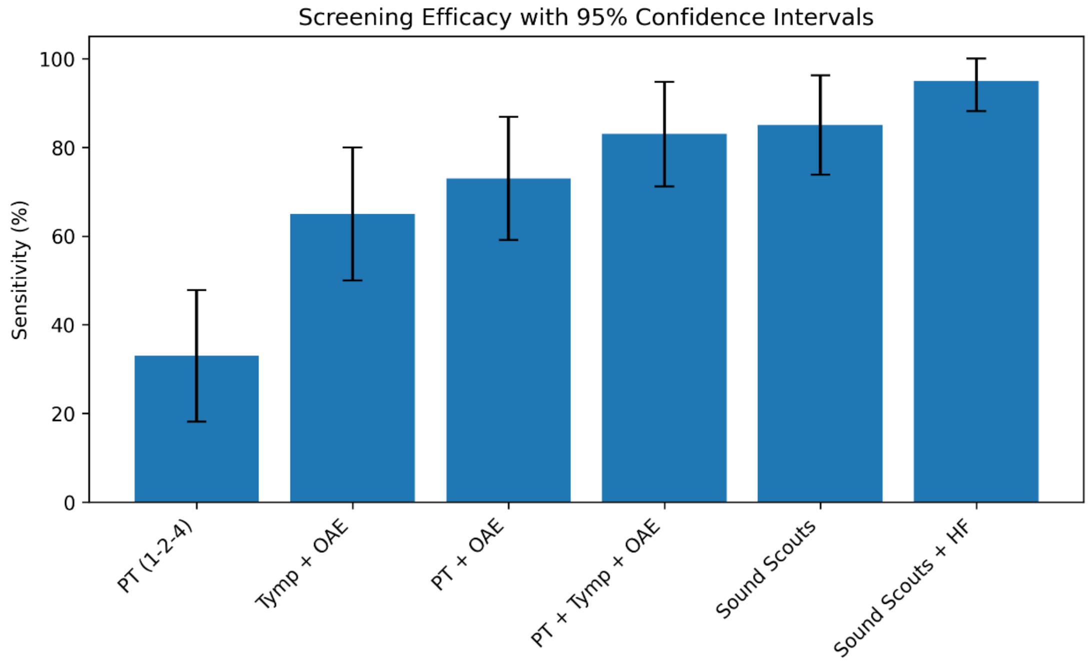
Abstract

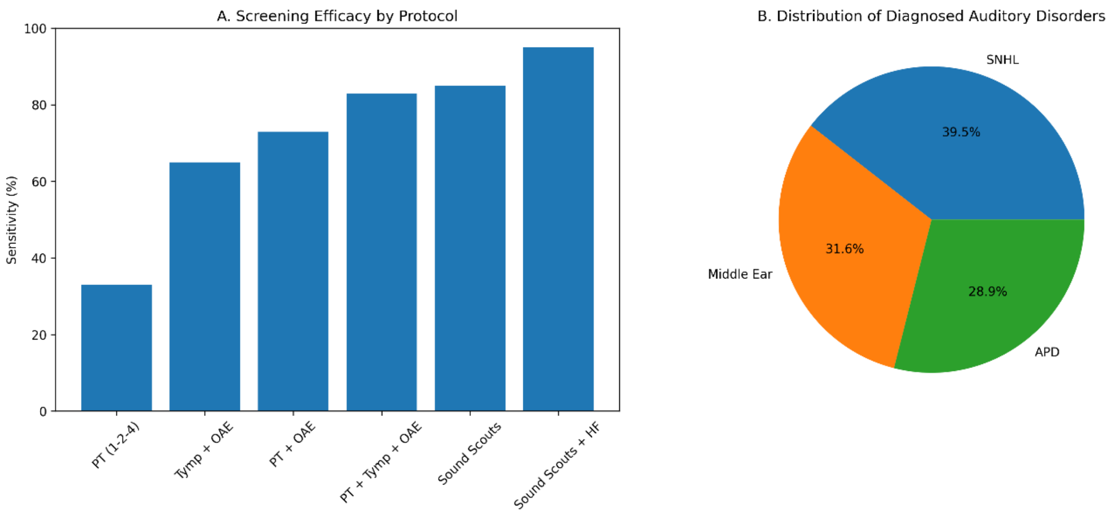
Background. School-based hearing screening relies on pure-tone detection, which does not assess functional listening. Many children with normal thresholds have difficulty understanding speech in noise, suggesting auditory processing or neural encoding deficits. This study evaluates speech-in-noise screening efficiency and explores electrophysiologic correlates in a subset completing follow-up, while examining rhythmic processing as a marker of auditory timing.

Methods. Children aged 5–9 years underwent screening including pure-tone audiometry, tympanometry, otoacoustic emissions, speech-in-noise assessment, and a tablet-based rhythmic synchronization task. Those who failed were referred for diagnostics. A subset completing follow-up underwent Auditory Brainstem Response and Frequency-Following Response testing and auditory processing measures. Analyses evaluated wave morphology, latency, interpeak intervals, and neural phase-locking.

Results. Speech-in-noise screening showed greater sensitivity for identifying functional deficits than pure-tone measures. Many children who passed traditional screening had difficulty in noise and were later diagnosed with auditory pathway disorders. In the electrophysiologic subset, findings included reduced Wave I amplitude, prolonged interpeak latencies, and degraded neural encoding of speech. Rhythmicity showed increased auditory timing variability but no robust group differences.

Conclusions. Speech-in-noise screening is an efficient front-line tool for identifying functional auditory deficits. Electrophysiologic findings support a neural basis for these difficulties, while rhythmic processing shows promise but requires further validation. Low follow-up rates reflect real-world barriers, underscoring the need for efficient screening tools to identify at-risk children early. Access to sound does not ensure access to information, particularly in complex listening environments.


Screening Efficacy with 95% Confidence Intervals
Sensitivity (%)
0
20
40
60
80
100
PT (1-2-4)
Tymp + OAE
PT + OAE
PT + Tymp + OAE
Sound Scouts
Sound Scouts + HF


A. Screening Efficacy by Protocol
Sensitivity (%)
0
20
40
60
80
100
PT (1-2-4)
Tymp + OAE
PT + OAE
PT + Tymp + OAE
Sound Scouts
Sound Scouts + HF
B. Distribution of Diagnosed Auditory Disorders
SNHL
39.5%
Middle Ear
31.6%
APD
28.9%

Data Modelling for openEHR Data Standards in Audiology: Archetypes for three typical Audiological Tests

Lena Schell-Majoor (Medical Physics, Big Data in Medicine and Cluster of Excellence Hearing4all.connects, Carl von Ossietzky Universität Oldenburg, Oldenburg, Germany); Mareike Buhl (Medical Physics and Cluster of Excellence Hearing4all.connects, Carl von Ossietzky Universität Oldenburg, Oldenburg, Germany; Institut de l'Audition and IHU reConnect, Paris, France)*; Eugen Kludt (Department of Otorhinolaryngology and Cluster of Excellence Hearing4all.connects, Medizinische Hochschule Hannover, Hannover, Germany); Daniel Berg (Medical Physics and Cluster of Excellence Hearing4all.connects, Carl von Ossietzky Universität Oldenburg, Oldenburg, Germany); Birger Kollmeier (Medical Physics and Cluster of Excellence Hearing4all.connects, Carl von Ossietzky Universität Oldenburg, Oldenburg, Germany); Antje Wulff (Big Data in Medicine and Cluster of Excellence Hearing4all.connects, Carl von Ossietzky Universität Oldenburg, Oldenburg, Germany)

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Poster

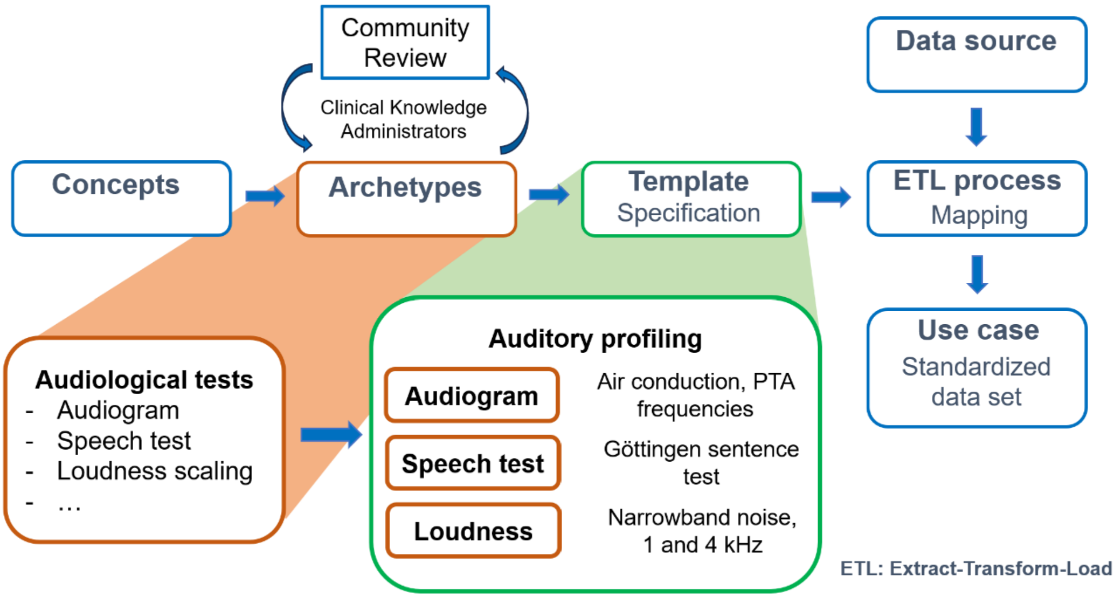
Abstract

Background. Analyses of combined, diverse audiological data from multiple locations are becoming increasingly important, for example in the context of AI-based decision support systems, or multi-center studies. To facilitate cross-institutional exchange, the efficient use of data and the combination of different data sources, standardization of audiological data is essential. An EFAS working group is working on the standardization of audiological data to advance data-driven audiology. The critical issue is data modelling, i.e., to determine which parameters need to be documented in which way in order to fully "understand" any new data set in a comprehensive, expert-knowledge-driven way that will allow to convert the provided data set into any standardized data format.

Methods. openEHR is an open interoperability standard that allows medical concepts to be described and stored in an unambiguous syntactic and semantic manner. Audiological tests are modelled using so-called archetypes, which can later be combined to templates (Fig. 1) to suit various use cases. Archetypes define which data elements are relevant to the respective clinical concept and describe the available data and metadata as completely as possible. To evaluate and optimize content and structure of the archetypes, a review was conducted with the audiological community.

Results. The specifities and output of the specific audiological modelling process are presented and discussed. We show the current audiological openEHR archetypes for loudness scaling, speech tests, and audiogram, which are published, under revision, and under development, respectively.

Conclusions. In summary, openEHR archetypes are proposed for three typical audiological tests and can be further refined through additional community review input. It is important to incorporate perspectives from clinical practice, research, and data structures even during the modelling phase, to achieve widely accepted data standards.



Hearing Data: Unlocking the potential of the Noah database

Nicolas Vannson (Hearing Data Science)*

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Poster

Abstract

Background. In France, every hearing aid clinic relies on the Noah database to store essential patient data, such as audiograms, hearing aid selection and fitting records. However, as this database grows daily, the lack of dedicated analytical tools limits its utility for patient follow-up and clinical insights.

Method: To address this gap, Hearing Data Science developed Hearing Data, a data-driven tool that unlocks the full potential of the Noah database to enhance patient care and clinical research. This tool currently offers four core features:

• Population overview: Age-categorized demographic insights

• Hearing aid analytics: Comprehensive analysis of fitting sessions, brands, models, and temporal trends

• Geodata integration: Geographical and socioeconomic profiling of patients

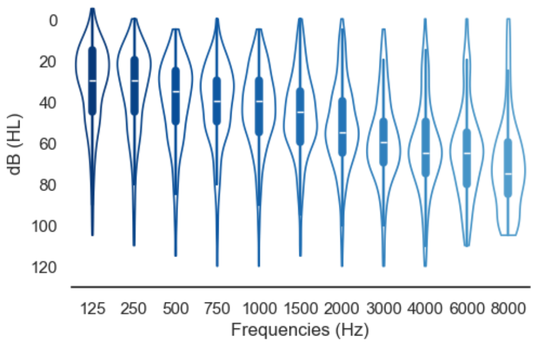
• Audiogram classifications: Standardized categorization using BIAP recommendations, Bisgaard classifications, and detailed hearing measurements

Results. Hearing Data has demonstrated high efficiency analyzing datasets of over 300 subjects per file in just 10–30 seconds. Ongoing development aims to aggregate data from multiple hearing clinics, further expanding its analytical power and scalability.

Conclusion: This brand-new tool is also available for research projects, offering a powerful resource for data-driven advancements in audiology.

Hearing Data Science Audiological stats for 278 patient(s)

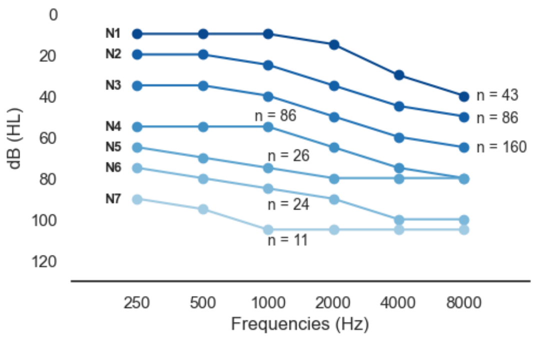
Unaided audiograms (right and left ears mixed)



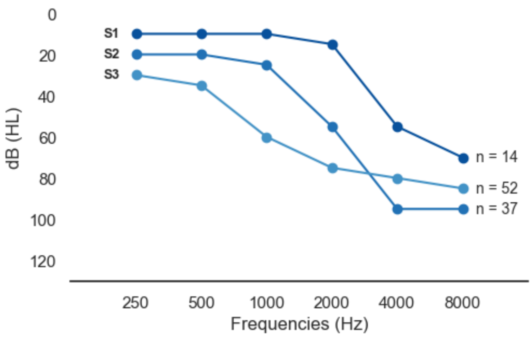
BIAP PTA $_{0.5,\,1,\,2,\,4\ kHz}$

BIAP	Hearing loss	Count
41-55	Moderate grade I	201
21-40	Mild	128
56-70	Moderate grade II	116
<=20	Normal	34
81-90	Severe grade II	24
71-80	Severe grade I	23
91-100	Profound grade I	6
111-119	Profound grade III	4
101-110	Profound grade II	3

Bisgaard audiogram classification (Flat)



Bisgaard audiogram classification (Steep)



Session 6.B. Multisensory and Vestibular Integration

Balance, cross-modal perception, and earable technologies.

Chaired by Dr. Seba Ausili.

Audio-Visual Speech Recognition: Effects of Auditory Masking and Visual Interference for Healthy Young Adults

Emma Søndergaard Pedersen (Department of Clinical Research, University of Southern Denmark, Odense, Denmark)*; Lukas Jürgensen (Department of Clinical Research, University of Southern Denmark, Odense, Denmark); Søren Krogh Andersen (Department of Psychology, University of Southern Denmark, Odense, Denmark); Tobias Neher (Department of Clinical Research, University of Southern Denmark, Odense, Denmark)

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Featured talk

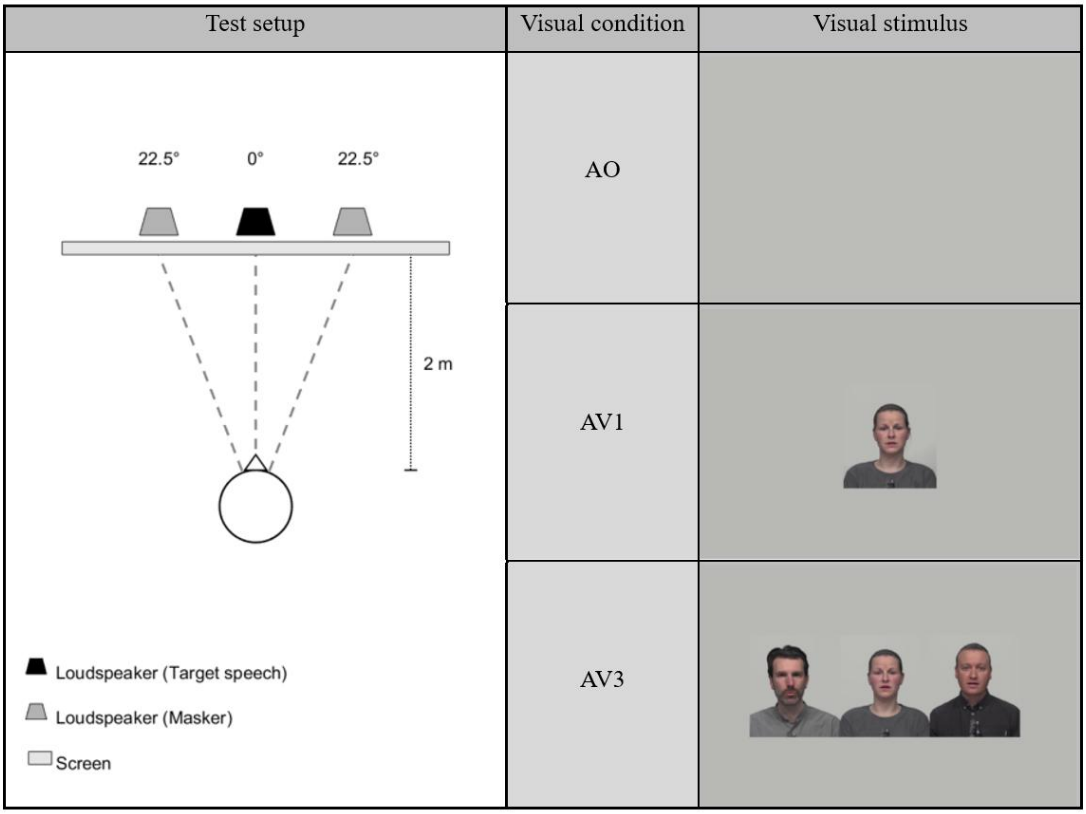
Abstract

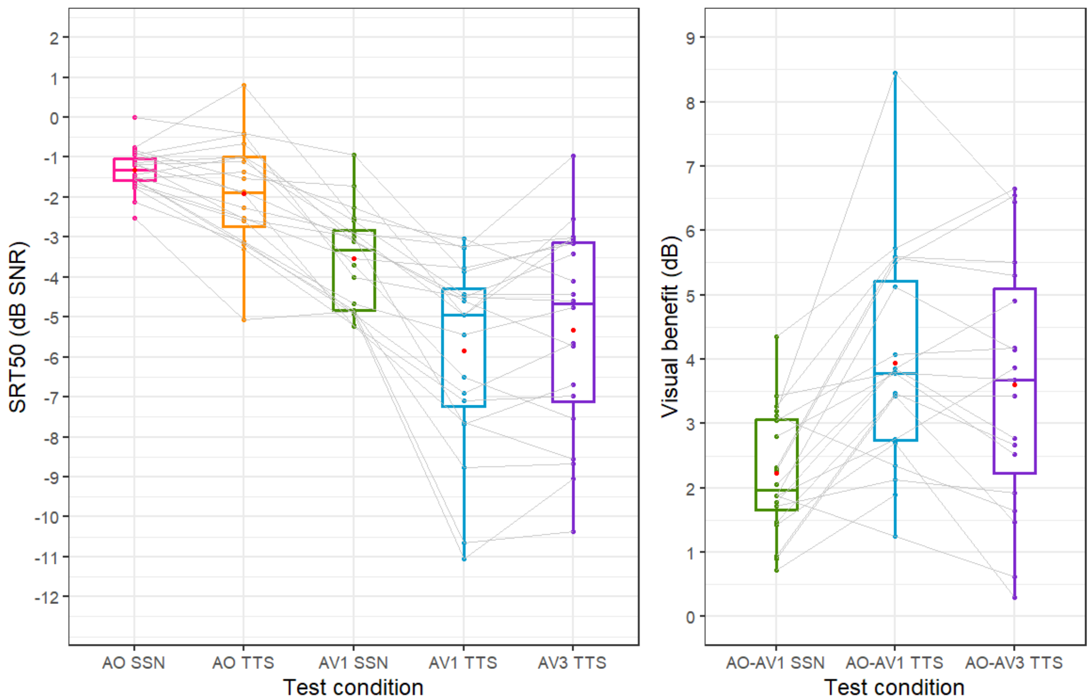
Background. Successful speech communication in complex environments requires identifying a target signal amid several competing signals. Acoustic and linguistic similarity of the target and competing signals can cause peripheral and central auditory masking. Visual target speech can improve identification, whereas visual competing speech can cause interference. This study investigated how different auditory maskers and visual conditions affect audio-visual speech recognition.

Method: Speech recognition thresholds (SRT50s) were measured using the 'Audio-Visual Danish Sentence Test' (AV-DAST). Twenty young adults with normal hearing and normal or corrected-to-normal vision participated. Five scenarios based on two auditory maskers (speech-shaped noise, SSN, and two-talker speech, TTS) and three visual conditions (audio-only, AO, visual target speech, AV1, visual target and competing speech, AV3) were tested. Visual (dis)benefit was calculated as the difference between corresponding SRT50 measurements (AO-AV1 and AV1-AV3). To evaluate test-retest reliability, all measurements were repeated after two weeks.

Results. With AO presentation, SSN and TTS gave comparable SRT50s (means: -1.3 dB SNR vs. -1.9 dB SNR). With AV1 presentation, TTS gave lower SRT50s than SSN (means: -5.8 dB SNR vs. -3.5 dB SNR), resulting in larger lip-reading benefit (AO-AV1) with TTS than with SSN (means: 3.9 dB vs. 2.2 dB). Visual competing speech resulted in a small overall disbenefit (AV1-AV3; mean: -0.5 dB) and large between-subject variability. Test-retest reliability of the various measurements was generally high.

Conclusions. Visual target speech improves speech recognition more in the presence of competing speech than in the presence of stationary noise. Young healthy adults differ markedly in their susceptibility to visual competing speech. The collected data provide a useful basis for follow-up work with participants with sensory impairments.


Test setup
Visual condition
Visual stimulus
22.5°
0°
22.5°
2 m
Loudspeaker (Target speech)
Loudspeaker (Masker)
Screen
AO
AV1
AV3


SRT50 (dB SNR)
Test condition
AO SSN
AO TTS
AV1 SSN
AV1 TTS
AV3 TTS
Visual benefit (dB)
Test condition
AO-AV1 SSN
AO-AV1 TTS
AO-AV3 TTS

Exploring the Auditory-Tactile Integration of Complex Acoustic Features using the Multichannel Vibrotactile Gloves

Loonan Chauvette (CERVO Brain Research Center)*; Anne Sophie Grenier (CERVO Brain Research Center); Emily Coffey (Concordia University); Robert Zatorre (Montreal Neurological Institute); Philippe Albouy (CERVO Brain Research Center); Jérémie Voix (École de technologie supérieure); Andréanne Sharp (CERVO Brain Research Center)

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Podium

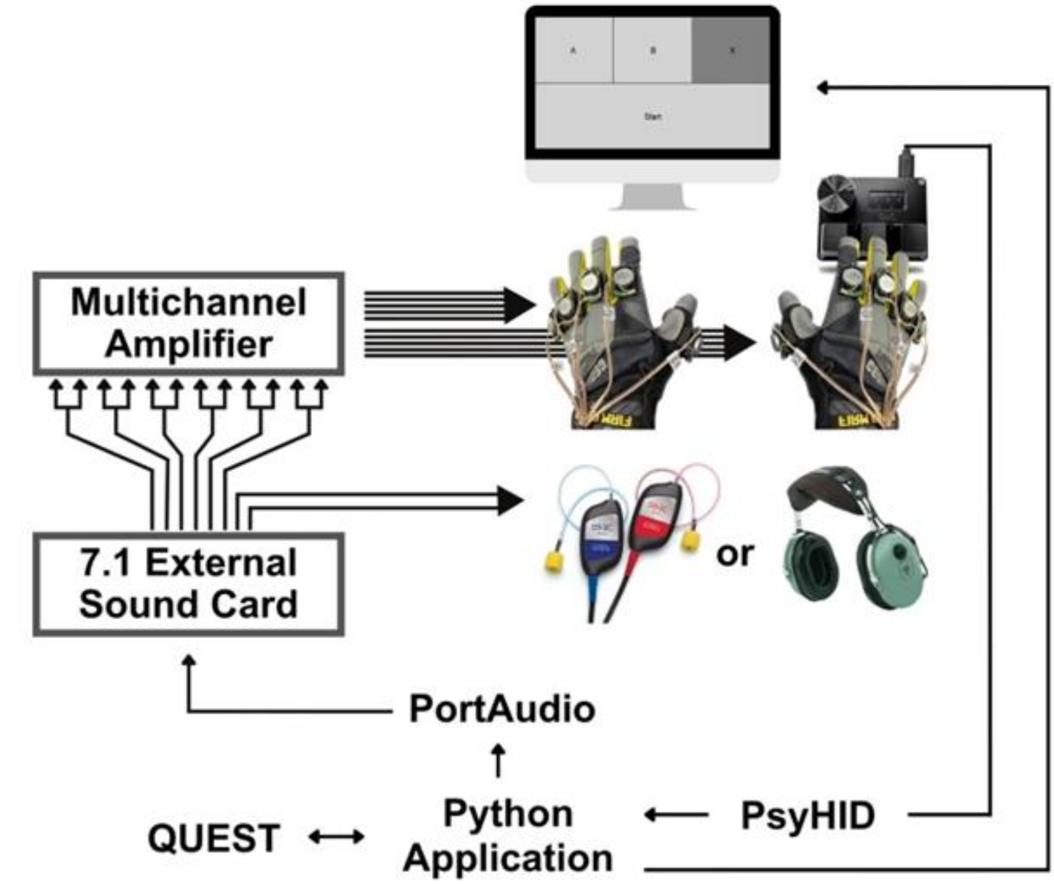
Abstract

Background. Auditory–tactile assistive technologies can complement conventional hearing devices, notably for music perception and noisy listening environments. However, designing effective devices requires knowledge of how acoustic cues are encoded and integrated across sensory modalities. Timbre perception, emerging from complex spectral and temporal acoustic cues, provides a framework to examine the limits and benefits of multisensory augmentation for complex sound perception.

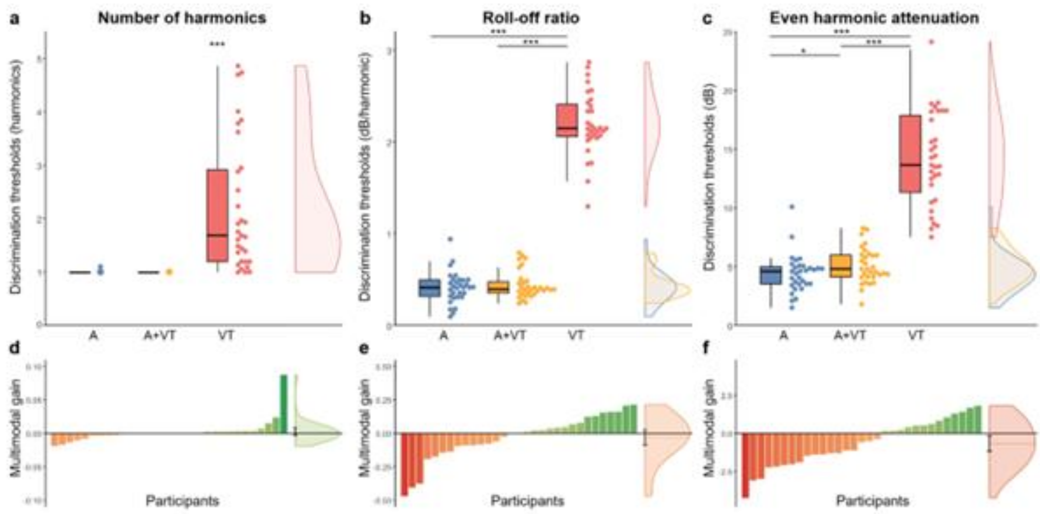
Methods. 30 adults with normal hearing and screened for vibrotactile acuity completed a timbre discrimination task in auditory, tactile, and auditory–tactile conditions. Stimuli were presented via insert earphones and the Multichannel Vibrotactile Glove, an assistive device we developed to convey acoustic signals as vibrations with 5 independent actuators per hand. Discrimination thresholds were estimated using an adaptive staircase method for 6 timbre-related acoustic parameters: 3 spectral (number of harmonics, harmonic roll-off, even-harmonic attenuation) and 3 temporal (attack time, amplitude modulation depth, modulation frequency).

Results. All timbre-related parameters could be discriminated using vibrotactile stimulation alone, indicating that complex acoustic cues can be conveyed through touch. Tactile thresholds were worse than auditory ones for spectral parameters but were comparable across modalities for temporal parameters. Auditory-tactile performance was statistically better for two out of three temporal features, suggesting cross-modal integration when unimodal cues share similar baseline sensitivity.

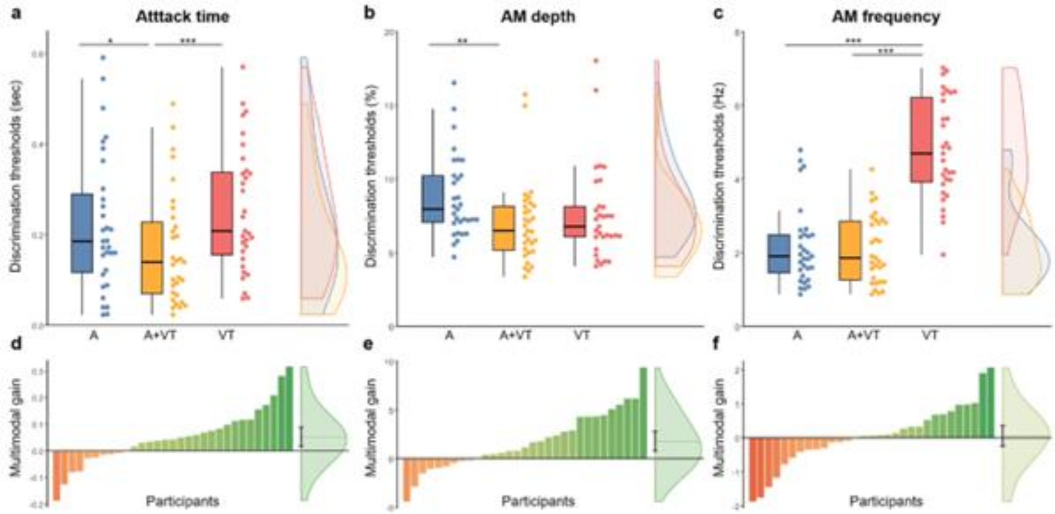
Conclusions. The study reveals constraints and opportunities for multisensory technologies. Vibrotactile augmentation seems best suited for temporal acoustic features, and multisensory gains depend on cross-modal threshold alignment. The Multichannel Vibrotactile Glove provides a flexible platform to explore signal processing strategies for multisensory hearing rehabilitation.



Experimental setup and apparatus used for the experiments.



Discrimination thresholds and multimodal gain for spectral acoustic features. The top plots (a-c) show the discrimination thresholds as boxplots, individual datapoints, and distributions for auditory (A) in blue, auditory + vibrotactile (A + VT) in yellow, and vibrotactile (VT) in red stimulation conditions. Lower thresholds indicate better discrimination performance. The bottom plots (d-f) show each participant's multimodal gain, calculated as their auditory thresholds minus their auditory + vibrotactile thresholds (A minus A + VT), in ascending order. Negative values (in red) represent a lower performance in the multimodal condition, while positive values (in green) represent better multimodal performance. The distributions on the right show the mean multimodal gain as the dotted line, with the error bar representing the 95% CI. *p <.05. **p <.01. ***p <.001.



Discrimination thresholds and multimodal gain for temporal acoustic features. The top plots (**a-c**) show the discrimination thresholds as boxplots, individual datapoints, and distributions for auditory (A) in blue, auditory + vibrotactile (A + VT) in yellow, and vibrotactile (VT) in red stimulation conditions. Lower thresholds indicate better discrimination performance. The bottom plots (**d-f**) show each participant's multimodal gain, calculated as their auditory thresholds minus their auditory + vibrotactile thresholds (A minus A + VT), in ascending order. Negative values (in red) represent a lower performance in the multimodal condition, while positive values (in green) represent better multimodal performance. The distributions on the right show the mean multimodal gain as the dotted line, with the error bar representing the 95% CI. *p <.05. **p <.01. ***p <.001.

The Effects of Cochlear Implants on the Vestibular System

Christian Steyn (Bioengineering, Department of Electrical, Electronic, and Computer Engineering, University of Pretoria)*; Tania Hanekom (Bioengineering, Department of Electrical, Electronic, and Computer Engineering, University of Pretoria)

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Podium

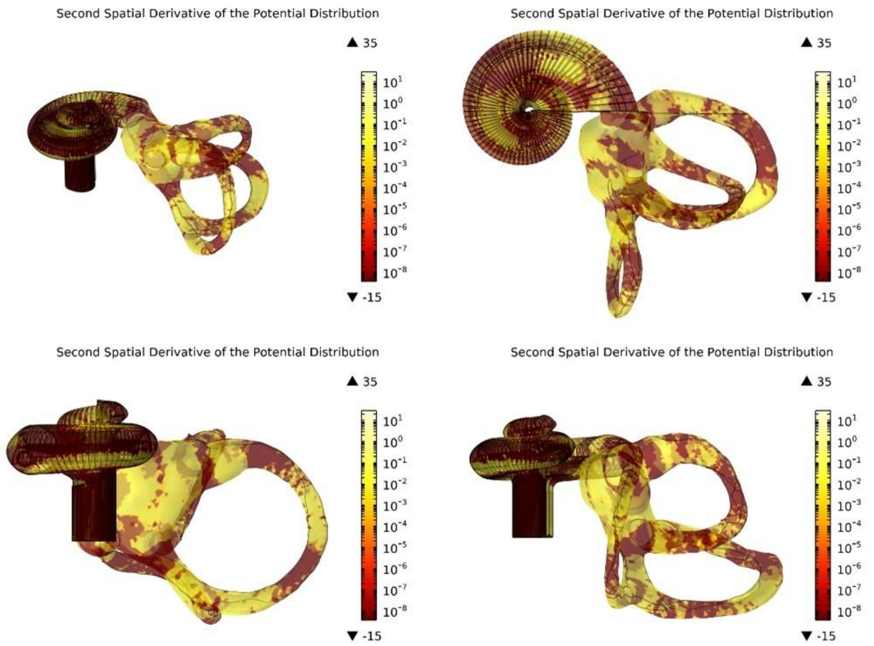
Abstract

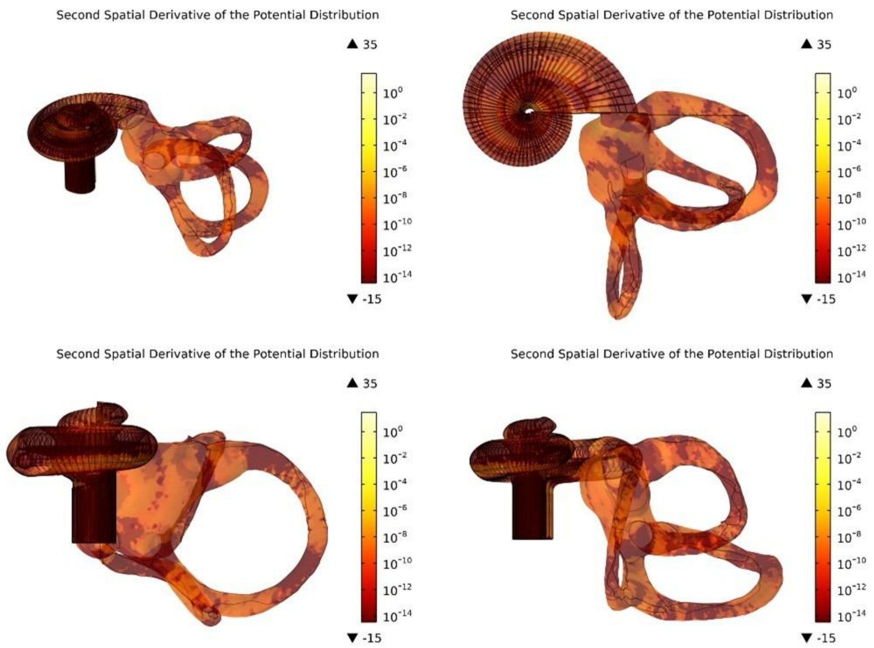
Background. Cochlear implants stimulate the cochlear nerve by injecting a current from an electrode located inside the cochlea. This current can spread throughout the inner ear and cause unintended stimulation of the facial nerve and the vestibular system. This study aims to understand the mechanism behind vestibular co-stimulation in cochlear implants users using a computational model.

Methods. A person-specific, three-dimensional finite element model of the inner ear is used to simulate the electric potential distribution in the inner ear of cochlear implant recipients. The model includes a detailed description of the vestibular system. The second spatial derivative of the resulting potential distribution, which is proportional to Rattay's activating function, is used as a metric for the probability of neural excitation throughout the cochleo-vestibular volume.

Results. Preliminary predictions of the activating function suggest that structures outside the cochlea are affected by intracochlear stimulation. Hotspots, where stimulation is likely, are visible near all five vestibular organs: the saccule; utricle; and the three ampullae.

Conclusions. The hotspots in the vestibular system suggest that all the vestibular organs could potentially be stimulated by the cochlear implant. Bipolar (apical reference) stimulation limits the occurrence of excitation hotspots outside the cochlea, whereas the model predicts a wider distribution of possible extracochlear excitation for monopolar stimulation. This observation is consistent with previous studies performed on spread of excitation to other extracochlear structures, such as the facial nerve.


Second Spatial Derivative of the Potential Distribution
▲ 35
10^{1} 10^{0} 10^{-1} 10^{-2} 10^{-3} 10^{-4} 10^{-5} 10^{-6} 10^{-7} 10^{-8}
▼ -15
Second Spatial Derivative of the Potential Distribution
▲ 35
10^{1} 10^{0} 10^{-1} 10^{-2} 10^{-3} 10^{-4} 10^{-5} 10^{-6} 10^{-7} 10^{-8}
▼ -15
Second Spatial Derivative of the Potential Distribution
▲ 35
10^{1} 10^{0} 10^{-1} 10^{-2} 10^{-3} 10^{-4} 10^{-5} 10^{-6} 10^{-7} 10^{-8}
▼ -15
Second Spatial Derivative of the Potential Distribution
▲ 35
10^{1} 10^{0} 10^{-1} 10^{-2} 10^{-3} 10^{-4} 10^{-5} 10^{-6} 10^{-7} 10^{-8}
▼ -15


Second Spatial Derivative of the Potential Distribution
▲ 35
10^{0} 10^{-2} 10^{-4} 10^{-6} 10^{-8} 10^{-10} 10^{-12} 10^{-14}
▼ -15
Second Spatial Derivative of the Potential Distribution
▲ 35
10^{0} 10^{-2} 10^{-4} 10^{-6} 10^{-8} 10^{-10} 10^{-12} 10^{-14}
▼ -15
Second Spatial Derivative of the Potential Distribution
▲ 35
10^{0} 10^{-2} 10^{-4} 10^{-6} 10^{-8} 10^{-10} 10^{-12} 10^{-14}
▼ -15
Second Spatial Derivative of the Potential Distribution
▲ 35
10^{0} 10^{-2} 10^{-4} 10^{-6} 10^{-8} 10^{-10} 10^{-12} 10^{-14}
▼ -15

Effects of Chiropractic Interventions on the Audio-Vestibular System: A Systematic Review

Şule Çekiç (Ankara Yıldırım Beyazıt University); Beyza Yılmaz (Ankara Yildirim Beyazit Üniversitesi)*; Sinem Özbey (Ankara Yildırim Beyazit University); Buse Çöllü (Ankara Yıldırım Beyazıt Üniversitesi); Eda Lale Köroğlu (Ankara Yildirim Beyazit University)

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Podium

Abstract

Objective. This study aims to investigate the effects of chiropractic interventions—a practice that has gained significant popularity—on the vestibular and auditory systems and to evaluate the clinical outcomes observed in patients.

Methods. A systematic search was conducted on November 31, 2024, across the PubMed, EBSCO, and Web of Science (WoS) databases following the PRISMA flow diagram. The search strategy utilized the following keyword combinations: (Chiropractic) AND (Hearing), (Chiropractic) AND (Vestibular), (Chiropractic) AND (Hearing loss), and (Chiropractic) AND (Vestibular system). PROSPERO: CRD42024615753

Results. Due to the prevalence of case reports, this review specifically details five studies (n=186; ages 0–75) consisting of case series, pilot, and non-randomized trials. Four of these studies reported symptomatic improvement following cervical-focused interventions, while one observed deterioration. Regarding clinical focus, two studies examined the vestibular system, one investigated the auditory system, and three addressed combined audio-vestibular disorders.

Discussion and Conclusion: The predominance of case reports over controlled trials highlights a gap in literature regarding chiropractic care's role in audio-vestibular health. Efficacy is closely linked to cervical sensorimotor control and musculoskeletal issues in the head-neck or temporomandibular regions. From an audiological perspective, distinguishing specific vestibular functions from general balance is critical. Therefore, a multidisciplinary approach—integrating objective tools like X-ray and MRI with otological expertise—is essential. Future research should utilize the localized clinical data presented herein to guide standardized clinical trials.

Keywords: auditory system, chiropractic, chiropractic interventions, vestibular system, tinnitus, audiovestibular system, systematic review

The Digital Triad: Screen Time, Sleep Disruption, and Vestibulo-Cochlear Dysfunction -A Pilot Study

Jerlin Glites (Noorul Islam Centre for Higher Education)*; Reshma Nair (Noorul Islam Centre for Higher Education)

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Poster

Abstract

Background. Prolonged screen time in young adults is linked to vestibular problems, cochlear issues, and disturbed sleep patterns, yet little is known about how these systems are convergently affected; this study investigates whether excessive screen time produces a consistent pattern of dysfunction across all three domains and their interrelationships.

Method: 36 young adults (18–25 years) were divided into high (>4 hrs/day, n=18) and low (<4 hrs/day, n=18) screen time groups, with screen time quantified via structured interview using a weighted daily average formula. Sleep quality, vestibular symptoms, and cochlear outer hair cell function were assessed using the Sleep Quality Scale (SQS), Vertigo Symptom Scale – Short Form (VSS-SF), and Distortion-Product OtoAcoustic Emissions DPOAEs (356–5645 Hz) respectively, with Shapiro-Wilk, Mann–Whitney U, independent samples t-tests, and Spearman's correlation employed for statistical analysis.

Results. The high screen time group showed significantly poorer sleep quality [$t = -4.29$, $p < .001$] and greater vestibular symptom severity [$U = 82.5$, $p = 0.012$] compared to controls, while DPOAE amplitudes revealed no significant difference between groups [$p > 0.05$]. Screen time was strongly correlated with poorer sleep quality [$\rho = 0.571$, $p < .001$] and moderately correlated with vestibular symptoms [$\rho = 0.470$, $p = .004$], with sleep quality and vestibular scores also significantly interrelated [$\rho = 0.333$, $p = .04$].

Conclusions. Excessive screen time negatively impacts sleep quality and vestibular function in young adults, with significant associations observed across these systems, while cochlear function remained unaffected, highlighting the need for longitudinal research to establish causality.

Average Screen Time (Hrs/Day) by Group

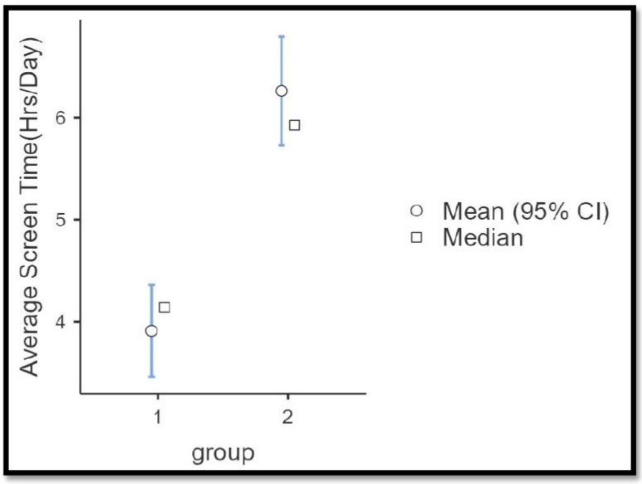

Figure 1. Dot plot showing mean and median average screen time (hrs/day) across two groups.

VSS Score by Group

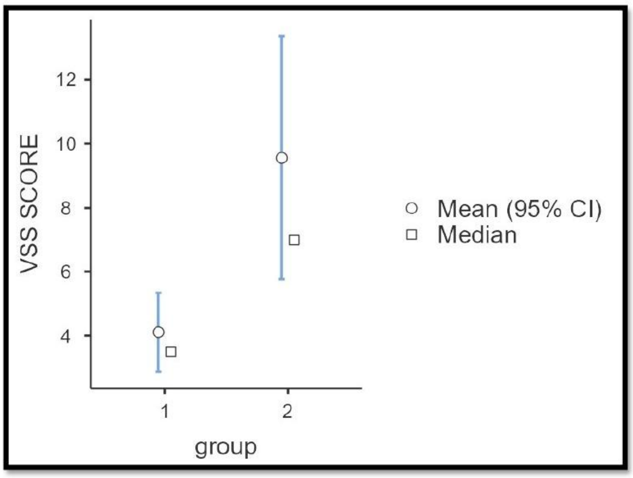

Figure 2. Dot plot showing mean and median VSS scores across two groups

SQS Score by Group

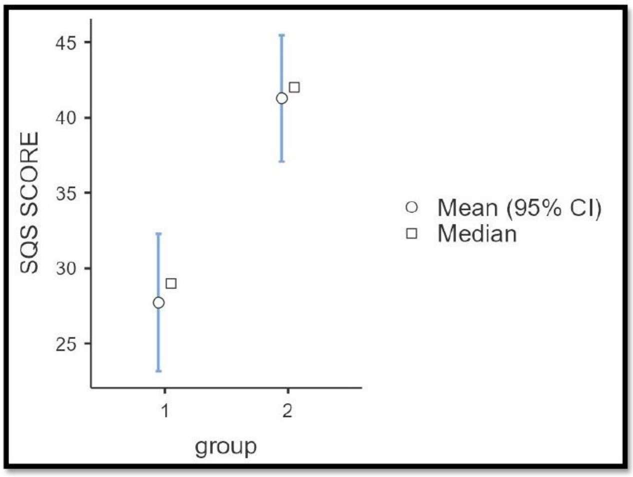

Figure 3. Dot plot showing mean and median SQS scores across two groups.

Multisensory Balance Assessment in Patients with Muscle Tension Dysphonia

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Poster

Abstract

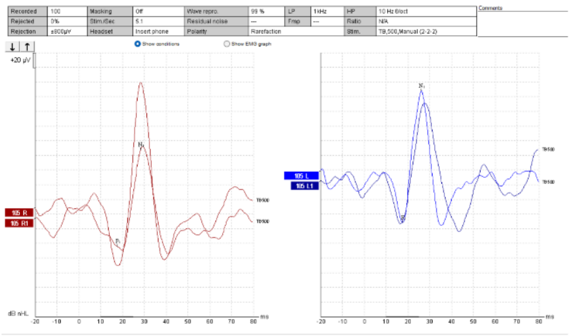
Background. Effective vocal production requires a healthy respiratory system, coordinated muscle activity, and proper posture. Since 2010, several studies have been published examining the connections between voice disorders and postural stability, however the literature is limited. The aim of this study is to examine the balance system of patients with muscle tension dysphonia (MTD) using posturography on a multisensory basis.

Methods. We evaluated the vestibular systems of patients over 18 years old diagnosed with vocal pathology using virtualis posturography and VEMP tests. We used computerized dynamic posturography so we temporarily removed or challenged specific parts of the balance system and analyzed the results to identify weaknesses in different components of postural control. We also used VEMP to measure cervical and ocular myogenic potentials. Since vocal pathology may cause tension in the neck muscles, we specifically evaluated cVEMP responses, which reflect activity in the neck muscles.

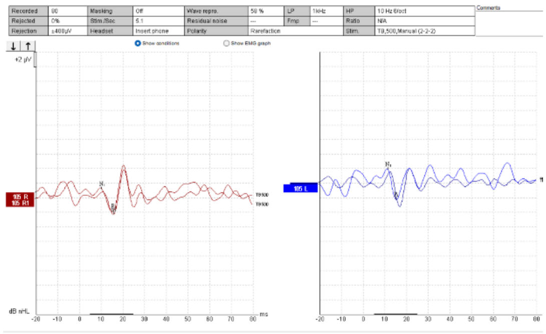
Results. Our study is being conducted as a pilot study, and two individuals diagnosed with MTD have been included in the research. In Sensory Organization Test weakness in the visual and vestibular systems was detected in the individuals included in the study. Adaptation Test results of the individuals were observed to be within normal limits, and they maintained their balance by developing coping strategies for repetitive movements. In Motor Control Test, abnormalities in latencies and amplitude scales were observed in both patients. According to the results, cVEMP response could not be obtained from the left side in one patient.

Conclusion. We found that the balance systems of individuals diagnosed with MTD were affected. Within the scope of the obtained data, we observed that the tension in the SCM muscle impacts the vestibular nerve and saccule function and that the impairment of the visual and vestibular systems leads to weakness in the balance system.

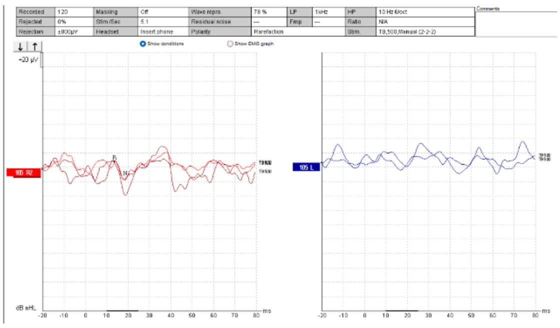
cVEMP



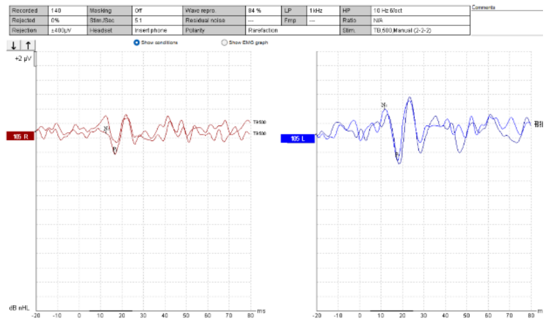
oVEMP



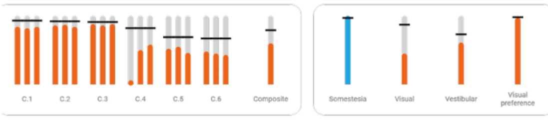
cVEMP



oVEMP

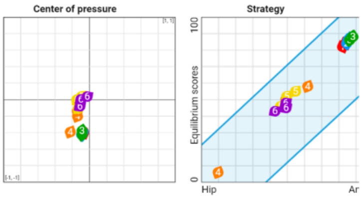


SOT (04/14/2026) – Results overview

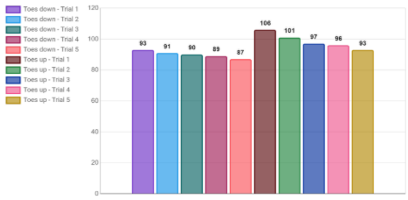


	Condition 1			Condition 2			Condition 3			Condition 4			Condition 5			Condition 6		
	1	2	3	1	2	3	1	2	3	1	2	3	1	2	3	1	2	3
Balance	84	82	84	86	87	84	88	86	88	6	50	58	52	55	46	48	45	43
- Norm	93			92			91			82			69			67		
Velocity (°/s)	4.9	5.1	5.1	5.0	5.2	5.2	5.0	5.0	5.0	8.2	5.8	5.5	6.1	5.5	5.7	5.7	5.4	5.9

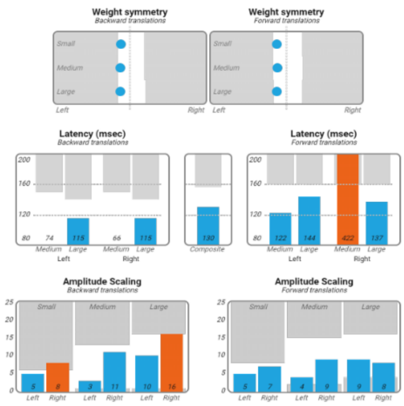
	Score	Norm
Composite	60	79
Somestesia	100	97
Visual	45	87
Vestibular	61	73
Visual preference	97	98



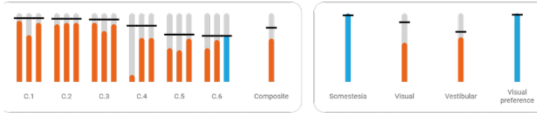
Adaptation Test (04/14/2026, 10:50 AM)



Motor Control Test (04/14/2026, 10:52 AM)

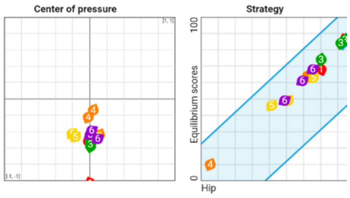


SOT (04/14/2026) – Results overview

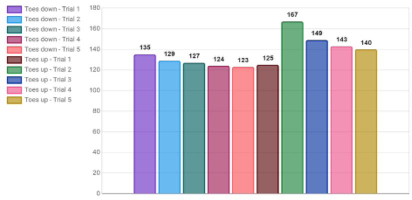


	Condition 1			Condition 2			Condition 3			Condition 4			Condition 5			Condition 6		
	1	2	3	1	2	3	1	2	3	1	2	3	1	2	3	1	2	3
Balance	89	68	86	84	86	86	86	74	84	10	64	64	49	46	63	49	61	68
- Norm	93			92			91			82			69			67		
Velocity (°/s)	5.6	5.1	4.9	5.1	5.0	5.0	5.0	4.9	5.1	9.3	5.2	5.2	5.7	6.7	5.1	5.8	5.5	5.[illegible]

	Score	Norm
Composite	63	79
Somestesia	100	97
Visual	57	87
Vestibular	65	73
Visual preference	100	98



Adaptation Test (04/14/2026, 03:06 PM)



Motor Control Test (04/14/2026, 03:10 PM)