2013 Conference on Implantable Auditory Prostheses

July 14-19, 2013

Granlibakken Conference Center

Lake Tahoe, CA

2013 CONFERENCE ON IMPLANTABLE AUDITORY PROSTHESES

July 14 -19, 2013

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Court 3
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Bearpaw
Deerwander
Pine Residence
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Treetop Adventure Park
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To Tahoe Rim Trail

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Attendee List..... .....to be added

PROGRAM OVERVIEW

Sunday July 14	
3:00PM - 10:00PM	Registration
7:00PM – Midnight	Welcome Reception
Monday July 16	
7:00AM	Breakfast opens
8:00AM – 11:00AM	Coding I: F0, Lexical Tones, Prosody, Coding II: Neural mechanisms and models
11:00AM – 12:45PM	Poster Viewing
12:00PM- 1:00PM	Lunch
1:30PM – 3:00PM	Young Investigator Mentoring Session
1:00PM – 5:00PM	Poster Viewing, free time
5:00PM – 7:00PM	Dinner
7:00PM – 9:00PM	Top-Down Processes, Plasticity
9:00PM – midnight	Poster Viewing and Social
Tuesday July 16	
7:00AM	Breakfast opens
8:35AM – 11:00AM	Development and Bilateral Implantation; Bimodal Hearing
11:00AM – 12:45PM	Poster Viewing
12:00PM- 1:00PM	Lunch
1:00PM - 3:00PM	Research Interface Workshop (Loizou PDA, HEINRI, Cochlear, MedEl, AB)
1:00PM – 5:00PM	Poster Viewing, free time
5:00PM – 7:00PM	Dinner
7:00PM – 9:00PM	Nerve Survival/Regeneration/Effects of Long-term Auditory Deprivation
9:00PM – midnight	Poster Viewing and Social
Wednesday July 17	
7:00AM	Breakfast opens
8:30AM –11:00AM	Cross-modal Plasticity and Development
11:00AM –12:45PM	Poster Viewing
12:00PM- 1:00PM	Lunch
1:00PM – 3:00PM	FDA and Company Workshop: New Technology
1:00PM – 5:00PM	Poster Viewing, free time
5:00PM – 7:00PM	Dinner
7:00PM-midnight	Social Event
Thursday July 18	
7:00AM	Breakfast opens
8:10AM – 11:00AM	Coding III: Signal Processing:Philipos Loizou Memorial Session
11:00AM – 12:45PM	Poster Viewing
12:00PM- 1:00PM	Lunch
1:00PM – 5:00PM	Poster Viewing, free time
5:00PM – 7:00PM	Dinner
7:00PM – 9:00PM	Coding IV: Spectral and Temporal Interactions
9:00PM – midnight	Poster Viewing and Social
Friday July 19	
7:00AM	Breakfast opens
9:00AM – 11:15AM	Novel Approaches: Other Fields
12:00PM-1:00PM	Lunch and Conference End
1:00PM	Buses leave for SFO

CIAP 2013 Speaker Program

Monday July 15: 8:00am – 11:00am

Coding I: F0, Lexical Tones, Prosody (Chair: Jan Wouters)

8:00am Welcome and Announcements: Chatterjee/Kral/Shannon

8:15 Li Xu: Cochlear Implants and Tonal Languages

8:45 XinLuo: Intrinsic and Extrinsic Pitch Cues to Tone Recognition with Cochlear Implants

9:15 Monita Chatterjee: Processing of Pitch, Prosody, Voice Emotion, and Lexical Tones by Cochlear Implanted Children

9:35 Break

Coding II: Neural mechanisms and models(Chair: Jan Wouters)

9:55 Bob Carlyon: Limitations of temporal pitch perception measured using the auditory nerve response to electrical stimulation

10:25 Jason Boulet: Facilitation, Accommodation and Refraction in Electrically-Stimulated Model Auditory Nerve Fibers Is Dependent on Location of Stimulation and Voltage-Gated Channel Type Distribution

10:40 Johan Frijins: Reducing Electrical Interaction during Parallel Stimulation Using Various Compensation Techniques

11:00am to 12:45pm: Poster viewing

12:00 noon to 1:00pm: Lunch

1:00pm Young Investigator Mentoring Session (Litovsky)

Monday July 15: 7:00pm – 9:00pm

Top-Down Processes, Plasticity (Chair: Bob Shannon)

7:00 David Pisoni: Some Observations on the Indexical Properties of Speech: What Are They and Why do We Need Them for Robust Speech Recognition and Spoken Language Understanding?

7:30 Ingrid Johnsrude: Effortful listening: perception and learning of degraded speech depends critically on attention

8:00 Break

8:15 Deniz Baskent: How cognitive factors can affect the outcome in cochlear implants

8:40 Lina Reiss: Pitch, Fusion, and Binaural Spectral Integration

9:00pm to Midnight: Poster and social session

Tuesday, July 16, 8:35am - 11:00am

Development and Bilateral Implantation; Bimodal Hearing (Chair: Bernhard Seeber)

8:35 Bertrand Delgutte: Why is binaural performance with bilateral cochlear implants poorer than expected? Insights from neural data

9:05 Ruth Litovsky: Emergence of Binaural Hearing in Bilaterally Implanted Children

9:35 Karen Gordon: Bilateral cochlear implantation can protect the developing auditory system from unilaterally driven reorganization

10:05 Break

10:25 Matthew Goupell: Spatial Attention in Bilateral Cochlear-Implant Users

10:45 Matthew Winn: Duplex Perception by Listeners with Bilateral Cochlear Implants: A Measure of Bilateral Integration of Speech
11:00am to 12:45pm: Poster viewing
12:00 noon to 1:00pm: Lunch
1:00pm to 3:00pm Research Interface Workshop (Loizou PDA, HEINRI, Cochlear, MedEl, AB)

Tuesday, July 16, 7:00pm to 9:00pm
Nerve Survival/Regeneration/Effects of Long-term Auditory Deprivation (Chair: Rob Shepherd)
7:00 Pat Leake: Long-Term Effects OfNeurotrophins And Electrical Stimulation On Survival And Function Of Cochlear Spiral Ganglion Neurons
7:30 John Middlebrooks: Loss of Temporal Fidelity in the Auditory Pathway Following Auditory Deprivation
8:00 Andrew Wise: Drug Delivery to the Cochlea for Neural Survival
8:30 Break
8:45 Bob Shannon/Karen Kirk: Tribute to Bill House
9:00pm to Midnight: Poster Viewing

Wednesday, July 17, 8:30am to 11:00am
Cross-modal Plasticity and Development (Chair: Mario Svirsky)
8:30 Anu Sharma: Changes in cortical resource allocation in hearing loss
9:00 Andrej Kral: Cross-Modal Plasticity in the Congenitally Deaf Cat
9:30 Break
9:50 Karen Kirk: Multimodal Sentence Recognition by Children with Cochlear Implants or Hearing Aids
10:20 Rebecca Dewey: Functional Near Infrared Spectroscopy Reveals Cortical Cross-Modal Reorganisation Following Deafness
10:35am to 12:45pm: Poster viewing
12:00 noon to 1:00pm: Lunch
1:00pm to 6:00pm: Poster Viewing, free time

Wednesday, July 17, 1:00pm to 3:00pm Afternoon Workshop
1:00 Shu-Chen Peng/Vasant Dasika: FDA process/protocols
1:30 Company Presentations on New Technology (20 min each: Advanced Bionics, Cochlear, Med-El)
2:30 Livermore Electrodes (Pannu)

Wednesday 7:00pm to Midnight: Dance Night

Thursday, July 18, 8:10am to 11:00am(Chair: Ruth Litovsky)

Coding III: Signal Processing:PhiliposLoizou Memorial Session

8:10 Michael Dorman: Philip Loizou – an Appreciation

8:30 Jan Wouters: Speech Processing Optimization in Cochlear Implants Based on Objective Measures

9:00 Qian-jie Fu: Signal Optimization for Cochlear Implants: Philip Loizou's Contribution to Cochlear Implant Research

9:30 Break

9:50 Tom Francart: Sound processing to improve transmission of interaural level differences with combined cochlear implant and hearing aid stimulation

10:10 Julie Bierer: Using the Estimated Electrode-Neuron Interface to Optimize Speech Processor Settings Improves Speech Perception

10:30 Yi Hu: Synthesis-Driven Noise Suppression Strategies Based on Harmonic Regeneration

10:45 Kostas Kokkinakis: Acoustic Reverberation in Cochlear Implant Processors with Two Microphones

11:00am to 12:45pm: Poster viewing

12:00 noon to 1:00pm: Lunch

Thursday, July 18, 7:00pm to 9:00pm

Coding IV: Spectral and Temporal Interactions (Chair: Johan Frijns)

7:00 Bryan Pfingst: Rehabilitation Strategies Based on Across-Site Patterns of Temporal Acuity

7:30 Colette McKay: Temporal information processing: can current focussing help?

8:00 Break

8:15 Charles Finley: Monopolar-Coupled, Ectopic Stimulation in Cochlear Implants

8:45 Rachel Scheperle: Relationships Among Peripheral and Central Electrophysiological Measures of Spatial / Spectral Resolution and Speech Perception in Cochlear Implant Users

9:00pm to Midnight: Poster Viewing

Friday, July 19, 9:00am to 11:15am

Novel Approaches: Other Fields (Chair: John Middlebrooks)

9:00 Announcements

9:15 Rob Shepherd: Visual prostheses: a progress report to the CIAP community

9:45 Tadayuki Nakagawa: Challenge for technological regeneration of the cochlear sensory epithelium: development of the artificial cochlear sensory epithelium using piezoelectric materials

10:15 Charley Della Santina: Progress Toward a Multichannel Vestibular Implant to Treat Profound Loss of Inner Ear Balance Sensation

10:45 Hannes Maier: Implantable Hearing Systems – Rationale, Concepts, Technology, Clinical Results and Perspectives

11:15 Check out and departure

12:00 noon to 1:00pm: Lunch 1:00pm Buses depart for SFO

POSTER SESSION M: MONDAY – 8AM TO MIDNIGHT

M1: MIXED STIMULATION RATES FOR ENCODING TEMPORAL CUES IN BILATERAL COCHLEAR IMPLANTS: Tyler Churchill, Alan Kan, Matthew Goupell, Ruth Litovsky

M2: BINAURAL SENSITIVITY OF BILATERAL COCHLEAR IMPLANTED PATIENTS TO AMPLITUDE MODULATED STIMULATION PRESENTED ON MULTIPLE ELECTRODES: Heath G. Jones, Alan Kan, Ruth Litovsky

M3: LISTENING IN A COCKTAIL PARTY:EVIDENCE FOR BINAURAL PROCESSING BY BILATERAL CI AND HEARING PRESERVATION LISTENERS : Louise Loiselle, Michael Dorman, William Yost, Sarah Cook

M4: WHICH CHILDREN SHOULD BE CONSIDERED CANDIDATES FOR BILATERAL COCHLEAR IMPLANTATION?: Rosemary E. S. Lovett, Quentin Summerfield, Deborah A. Vickers

M5: PITCH RANKING IN BILATERAL ASYMMETRIC COCHLEAR IMPLANT ELECTRODE ARRAYS: Tobias Rader, TimoStoever, Uwe Baumann

M6: AUDIO-VISUAL SPEECH INTELLIGIBILITY BENEFITS WITH BILATERAL COCHLEAR IMPLANTS WHEN TALKER LOCATION VARIES DYNAMICALLY: Richard J.M. van Hoesel

M7: MEASURING THE BILATERAL ADVANTAGE IN COCHLEAR IMPLANTEES USING A BLIND MODEL: Stefano Cosentino, Torsten Marquardt, and David McAlpine

M8: RELATION BETWEEN INTERAURAL ENVELOPE DELAY SENSITIVITY AND SPEECH ENVELOPE STATISTICS IN ELECTRIC STIMULATION: Bernhard Laback, PiotrMajdak, Katharina Egger

M9: IMPROVING LOCALIZATION BY PRESERVATION OF ILD CUES: Christopher J Long, Lakshmish Ramanna, Wendy S Parkinson, Zachary M Smith

M10: THE PRECEDENCE EFFECT: INSIGHTS FROM ELECTRIC HEARING: Andrew D. Brown, Heath G. Jones, Alan H. Kan, Matthew J. Goupell, Christopher Stecker, Ruth Y. Litovsky

M11: NEURAL CODING OF INTERAURAL LEVEL DIFFERENCES WITH BILATERAL COCHLEAR IMPLANTS: Yoojin Chung[1,2], Kenneth E. Hancock[1,2], and Bertrand Delgutte[1,2,3]

M12: PRE- AND POST-OPERATIVE BINAURAL UNMASKING FOR BIMODAL COCHLEAR-IMPLANT LISTENERS: Benjamin Sheffield, Gerald Schuchman, Joshua Bernstein

M13: BINAURAL SPEECH-STREAM SEGREGATION FOR SINGLE-SIDED DEAF AND BILATERAL COCHLEAR IMPLANTEES: Joshua G.W. Bernstein, Matthew J. Goupell, NandiniIyer, Gerald Schuchman, Arnaldo L. Rivera, Douglas S. Brungart

M14: BINAURAL UNMASKING WITH TEMPORAL FINE STRUCTURE IN COCHLEAR IMPLANT LISTENERS: Ann E. Todd, Matthew J. Goupell, Ruth Y. Litovsky

M15: WITHIN-SUBJECT COMPARISONS OF ELECTROPHYSIOLOGIC AND PSYCHOPHYSICAL MEASURES OF SPECTRAL RIPPLE DENSITY DISCRIMINATION IN CI USERS WITH SINGLE SIDED DEAFNESS: Nicholas P Giuliani, Carolyn Brown, Paul Abbas, Christine Etler

M16: BILATERAL CHANGES OF SPONTANEOUS ACTIVITY WITHIN THE CENTRAL AUDITORY PATHWAY UPON UNILATERAL INTRA-COCHLEAR ELECTRICAL STIMULATION: Dietmar Basta, Moritz Groeschel, Patrick Boyle, Rolf-Dieter Battmer, Arne Ernst

M17: THE CONTRIBUTION TO BIMODAL BENEFIT OF DIFFERENT LOW FREQUENCY REGIONS IN CONTRALATERAL RESIDUAL HEARING: Marine Ardoint, Stuart Rosen, Timothy Green, Andrew Faulkner

M18: COCHLEAR IMPLANTATION IN SINGLE SIDED DEAFNESS: BENEFITS IN SPEECH PERCEPTION AND SOUND SOURCE LOCALIZATION: Stefan Michael Brill

M19: AUDITORY EVOKED RESPONSES TO PITCH MATCHED STIMULI IN UNILATERAL COCHLEAR IMPLANT USERS WITH RESIDUAL HEARING IN THE CONTRALATERAL EAR: Chin-Tuan Tan, Brett Martin , Keena Seward, KseniaProsolovich, Benjamin Guo, Elizabeth Glassman, Mario Svirsky

M20: RATE OR PLACE: WHICH IS THE REAL PITCH?: David M Landsberger, Andrea Kleine-Punte, Paul Van De Heyning

M21: CONFLICTING CUE VOWEL IDENTIFICATION BY BIMODAL CI USERS AND BY NORMAL HEARING LISTENERS USING AN ACOUSTIC MODEL OF A CI AND CONTRALATERAL HEARING AID: Elad Sagi, Arlene C. Neuman, E. Katelyn Glassman, Keena P. Seward, Chin-Tuan Tan and Mario A. Svirsky

M22: EFFECTS OF CHRONIC ELECTRO-ACOUSTIC STIMULATION ON CORTICAL RESPONSES: James Fallon, Sam Irving, Andrew Wise, Robert Shepherd, Hugh McDermott, Dexter Irvine

M23: THE ROLE OF PITCH CUES FROM CONTRALATERAL RESIDUAL ACOUSTIC HEARING IN COCHLEAR IMPLANT USERS' PERCEPTION OF INTONATION: Tim Green, Andrew Faulkner, Stuart Rosen, Marine Ardoint

M24: COCHLEAR IMPLANTATION IN ADULTS WITH ACQUIRED SINGLE-SIDED DEAFNESS: A PROSPECTIVE UK TRIAL: Padraig Thomas Kitterick, Gerard M. O Donoghue, Douglas E. H. Hartley, Bernhard U. Seeber

M25: SPEECH RECOGNITION IN NOISE AND LOCALIZATION ABILITIES IN ADULTS WITH UNILATERAL HEARING LOSS: IMPLICATIONS FOR COCHLEAR IMPLANT CANDIDACY: Jill B. Firszt, Ruth M. Reeder, Laura K. Holden

M26: CORTICAL BINAURAL INTERACTIONS AND ITD REPRESENTATION OF CATS WITH CONGENITAL SINGLE-SIDED DEAFNESS: JochenTillein[1,3], Peter Hubka[1], Andrej Kral[2]

M27: ELIMINATION OF OUTER HAIR-CELL GAIN EXPLAINS THRESHOLD CHANGES IN COCHLEAR IMPLANTED EARS: Bernd Hose, Chris James, Ernst von Wallenberg, Henryk Skarzynski, Artur Lorens, Monika Matusiak, Marek Porowski, Piotr Skarzynski

M28: USING TACTILE AIDS TO PROVIDE LOW FREQUENCY INFORMATION FOR COCHLEAR IMPLANT USERS: Shuai Wang, XuanZhong, Michael Dorman, William Yost, Julie Liss

M29: MULTI-ELECTRODE LATERALIZATION USING INTERAURAL LEVEL DIFFERENCES IN BILATERAL COCHLEAR IMPLANT SUBJECTS: Olga Stakhovskaya, TanviThakkar, Matthew J Goupell

M30: RATIONALE AND IMPLEMENTATION OF A BIMODAL FITTING FORMULA: Josef Chalupper, Smita Agrawal, Stefan Fredelake, Tony Spahr

M31: SOME ACOUSTIC MODELS OF COCHLEAR IMPLANTS ARE QUITE GOOD BUT THE MOST COMMONLY USED ONES ARE NOT: A SINGLE-SIDED DEAFNESS STUDY: Mario Svirsky

M32: EFFECTS OF COCHLEAR IMPLANT LOWER CUTOFF FREQUENCY ON SPEECH PERCEPTION IN BIMODAL COCHLEAR IMPLANT USERS: Jennifer Fowler, Jessica Eggleston, Lina Reiss

M33: MULTISENSORY INTEGRATION IN ADULT COCHLEAR IMPLANT USERS WITH HEARING PRESERVATION: Sterling Sheffield, Rene Gifford, Ryan Stevenson, Juliane Krueger, Mark Wallace

M34: INVESTIGATION OF THE TOP-DOWN PROCESS IN SIMULATED COMBINED ELECTRIC AND ACOUSTIC STIMULATION: Ying-Yee Kong [1], AlaMullangi[1], and Gail Donaldson [2]

M35: DEVELOPMENT OF CORTICAL PROCESSING OF CHANGES IN PIANO TONES IN EARLY-IMPLANTED CHILDREN; CONNECTIONS TO SINGING AND PERCEPTION OF SPEECH IN NOISE: RitvaTorppa, Andrew Faulkner, Minna Huotilainen

M36: VISUALIZATION OF HUMAN INNER EAR ANATOMY WITH HIGH RESOLUTION 7 TESLA MAGNETIC RESONANCE IMAGING: MA vd Jagt[1], WM Brink[2,3], MJ Versluis[2,3], JJ Briaire[1], AG Webb[2,3], JHM Frijns[1,4], BM Verbist[2,5]

M37: SOURCE ANALYSIS OF EASSR GENERATORS MODULATED AT SPEECH ENVELOPE RATES: Robert Luke, Michael Hofmann, Lieselot Van Deun, Astrid van Wieringen, Jan Wouters

M38: ASSESSING THE EFFECTS OF DEAFNESS ON CORTICAL NETWORKS IN THE DEVELOPING HUMAN AUDITORY SYSTEM: Desiree Kartika de Vreede, Patrick Yoo, Blake C. Papsin, Karen A. Gordon

M39: MECHANISMS UNDERLYING HEARING PRESERVATION IN CI USERS: Samidha Joglekar, Monita Chatterjee, Lendra M Friesen

M40: EXPANDING COCHLEAR IMPLANT NEURAL RESPONSE MEASUREMENT CAPABILITIES TO CORTICAL POTENTIALS: Myles Mc Laughlin, Alejandro Lopez-Valdes, Richard B Reilly, Fan-Gang Zeng

M41: INVESTIGATING A NOVEL METHOD FOR BILATERAL COCHLEAR IMPLANT FITTING: SPEECH OUTCOMES AND ENHANCED ILD DISCRIMINATION: Wendy S. Parkinson, Zachary M. Smith

M42: MODELING SOUND LOCALIZATION IN COCHLEAR IMPLANTS: Christian Wirtz, Michele Nicoletti, Peter Nopp, Peter Schleich, Werner Hemmert

M43: AGE AT ONSET OF HEARING LOSS, BILATERAL COCHLEAR IMPLANT EXPERIENCE AND SPATIAL RELEASE FROM MASKING: Alan Kan, Matthew B Winn, Heath Jones, Ruth Y Litovsky

M44: EMERGENCE OF SPATIAL HEARING ACUITY IN YOUNG BILATERALLY IMPLANTED CHILDREN: Shelly P Godar, Ruth Y Litovsky

M45: PSYCHOACOUSTICAL AND SELF-REPORTED OUTCOMES OF A CO-ORDINATED BILATERAL COCHLEAR IMPLANT SYSTEM IN ADULT USERS: Noura Alothman, Daniel Rowan , Helen Cullington

M46: LOCALIZATION OF CONCURRENT SOUND SOURCES USING BILATERAL COCHLEAR IMPLANTS: INFORMATIONAL AND ENERGETIC CONSIDERATIONS: Aswin Wijetillake, Richard Van Hoesel, Robert Cowan

M47: SEQUENTIAL STREAM SEGREGATION IN BILATERAL COCHLEAR IMPLANT USERS: Aswin Wijetillake, Richard Van Hoesel, Robert Cowan

M48: BENEFIT OF CONTRALATERAL ROUTING OF SIGNALS FOR UNILATERAL CI USERS: Cees H. Taal[1], Denise C.P.B.M. van Barneveld[2], Jeroen J. Briaire[1], Johan H.M. Frijns[1]

M49: WIRELESS MULTI-MICROPHONE TECHNOLOGY AND ITS ADVANTAGES FOR BILATERAL COCHLEAR IMPLANTS USERS: Gunnar Geissler, Iris Arweiler, Karl-Heinz Dyballa, Stefan Fredelake, Phillipp Hehrmann, Volkmar Hamacher, Andreas Buechner

M50: IMPROVING COCHLEAR IMPLANT PATIENTS' SPECTRAL RESOLUTION BY DIVIDING THE SIGNAL ACROSS EARS: Justin M Aronoff, Monica Padilla, Qian-Jie Fu, David M Landsberger

M51: EFFECT OF PHONOLOGICAL ERRORS ON SPEECH INTELLIGIBILITY AND DURATION OF SENTENCES OF COCHLEAR IMPLANT USERS: Olga Peskova, Nirmal Kumar Srinivasan, Sujin Shin, Madhu Sundarrajan, Ann Geers, Emily Tobey

M52: LONG TERM RESULTS OF ABI IN CHILDREN: Liliana Colletti, Robert Shannon, Vittorio Colletti

M53: THE DILEMMA OF CND IN CHILDREN : CI OR ABI? PERSONAL EXPERIENCE ON 21 CHILDREN FITTED FIRST WITH CI AND SUBSEQUENTLY WITH ABI: Liliana Colletti, Eric Wilkinson, Vittorio Colletti

M54: NARRATIVE LANGUAGE SKILLS IN SCHOOL-AGED CHILDREN WITH COCHLEAR IMPLANTS: Tinne Boons, Leo De Raeve, MargreetLangereis, Jan Wouters, Astrid van Wieringen

M55: CHARACTERISING CI CHANNEL PERFORMANCE USING AUDITORY STEADY-STATE RESPONSES: Robert Luke, Michael Hofmann, Lieselot Van Deun, Astrid van Wieringen, Jan Wouters

M56: MASKING RELEASE IN SPEECH WITH DEGRADED SPECTRAL CUES: EVIDENCE FROM EVENT-RELATED POTENTIALS: Yingjiu Nie, Sharon E. Miller, Tess K. Koerner, Peggy B. Nelson, Yang Zhang

M57: A FLEXIBLE MONOPOLAR STIMULATOR FOR ANIMAL STUDIES IN AUDITORY PROSTHESES: Feng Hong, Douglas Kim, Hussnain Ali, Philipos Loizou, John Hansen, Emily Tobey

M58: REMOVABLE INSERTION SHANKS FOR PENETRATING AUDITORY IMPLANTS: Sarah H Felix, Heeral J Sheth, Kedar G Shah, Vanessa M Tolosa, Angela C Tooker, Terri L Delima, William Benett, Satinderpall S Pannu

M59: CHRONIC MICROELECTRODE SYSTEM FOR PENETRATING AUDITORY IMPLANTS: Vanessa Tolosa, Angela Tooker, Sarah Felix, Kedar Shah, Heeral Sheth, Terri Delima, Satinderpall Pannu

M60: CHRONIC PERCUTANEOUS ELECTRICAL CONNECTOR FOR ELECTRICAL RECORDING AND STIMULATION WITH MICROELECTRODE ARRAYS: Kedar Shah, William Benett, Terri DeLima, Sarah Felix, Heeral Sheth, Vanessa Tolosa, Angela Tooker,Satinderpall Pannu

POSTER SESSION T: TUESDAY–8AM TO MIDNIOGHT

T1: WHAT CAN WE EXPECT OF A CONTEMPORARY GROUP OF CHILDREN WITH COCHLEAR IMPLANTS REGARDING THEIR AUDITORY, LINGUISTIC AND COGNITIVE SKILLS?: Astrid van Wieringen, Tinne Boons, Jan Wouters

T2: SPECTRAL RIPPLE DISCRIMINATION IN INFANTS WITH IMMATURE OR ATYPICAL SPECTRAL RESOLUTION: David L Horn, Jong Ho Won, Rosemary E.S. Lovett, Vasant K Dasika, Jay T Rubinstein, Lynne A Werner

T3: PITCH PERCEPTION AND LINKS TO LANGUAGE DEVELOPMENT IN PRIMARY SCHOOL CHILDREN WITH COCHLEAR IMPLANTS: Lucy Carroll, Andrew Faulkner, Debi Vickers

T4: SENSITIVITY TO BINAURAL CUES IN NORMAL HEARING CHILDREN AND CHILDREN WHO USE BILATERAL COCHLEAR IMPLANTS: Erica Ehlers, Yi Zheng, Alan Kan, Shelly Godar, Ruth Litovsky

T5: COGNITIVE, COMMUNICATIVE AND PERCEPTIVE DEVELOPMENT AFTER THREE YEARS OF IMPLANTATION OF CHILDREN AGED 6 TO 9: Aurore Berland, Michele Guidetti, Bernard Fraysse, Nadine Cochard, Pascal Barone, Pascal Gaillard, Olivier Deguine

T6: COCHLEAR IMPLANTATION OF CHILDREN WITH STEEPLY SLOPING HEARING LOSS: Margret Meredith, Jay T Rubinstein, Kathleen CY Sie, Susan J Norton

T7: OBJECTIVE MEASUREMENTS OF BALANCE DYSFUNCTION IN CHILDREN WITH SENSORINEURAL HEARING LOSS AND COCHLEAR IMPLANTS: Sharon L Cushing, Nikolaus E Wolter, Blake C Papsin, Karen A Gordon

T8: AUTOMATED ECAP CLASSIFICATION: Idrick Akhoun, Patrick Boyle, Volkmar Hamacher

T9: PROGRESSIVE LOW FREQUENCY HEARING LOSS IN A GUINEA PIG MODEL OF COCHLEAR IMPLANTATION BY ALTERING ELECTRODE POSITION: David Rowe, Stephen O'Leary

T10: TARGET STRUCTURES FOR INFRARED NEURAL STIMULATION (INS): Hunter Kendall Young, Sama Kadakia, Claus-Peter Richter

T11: INVESTIGATING THE EFFECT OF COCHLEAR GEOMETRY AND NEURAL MODELS ON THRESHOLDS WHEN MODELLING THE COCHLEAE OF SPECIFIC IMPLANT USERS: Werner Badenhorst, Tiaan K Malherbe, Tania Hanekom, Johan J Hanekom

T12a: ECAP THREHOLDS IN COCHLEAR IMPLANT USERS MEASURED INTRAOPERATIVELY ARE HIGHER THAN THOSE MEASURED POSTOPERATIVELY: Lutz Gaertner, Andreas Buechner, Thomas Lenarz

T12b: MEASURING ELECTRICALLY EVOKED COMPOUND ACTION POTENTIALS AS A FITTING TOOL FOR ABI PATIENTS: Thilo Rode, Gert Joseph, Lutz Gaertner, Andreas Buechner, Thomas Lenarz

T13: MIDBRAIN ACTIVITY EVOKED BY PULSED LASER LIGHT: Peter Baumhoff, Michael Schultz, Nicole Kallweit, Alexander Krueger, Tammo Ripken, Thomas Lenarz, Andrej Kral

T14: FREQUENCY FOLLOWING RESPONSES TO VOCODED SPEECH: Saradha Ananthakrishnan, Xin Luo, Ananthanarayan Krishnan

T15: ENVELOPE SHAPE AFFECTS NEURAL ITD CODING WITH BILATERAL COCHLEAR IMPLANTS: Kenneth E. Hancock, Yoojin Chung, Bertrand Delgutte

T16: APPLICATION OF NEURAL NETWORKS IN PREDICTING SUCCESS WITH COCHLEAR IMPLANTS: Sabine Haumann, Thomas Lenarz, Andreas Buechner

T17: ANIMAL MODELS FOR COCHLEAR IMPLANTATION TO ASSESS COCHLEAR TRAUMA: David McAlpine, Lisa S Nolan, Nina Mistry, Shakeel Saeed, Andrew Forge, Ruth R Taylor

T18: SENSITIVITY OF COCHLEAR VOLUME CONDUCTION MODELS TO BOUNDARY CONDITIONS: Paul Wong, Phillip Tran, Andrian Sue, Qing Li, Paul Carter

T19: BIOCOMPATIBILITY OF DEGRADABLE POLYMERS FOR COATING OF THE COCHLEAR IMPLANT ELECTRODE ARRAY: Gerrit Paasche, Piera Ceschi, Anne Bohl, Thomas Lenarz, Katrin Sternberg

T20: POTENTIAL BENEFIT OF EMULATING THE COCHLEAR TRAVELLING WAVE IN COCHLEAR IMPLANT SPEECH PROCESSORS: Johan Hanekom, Larry Schmidt, Tiaan Du Preez

T21: A NOVEL, AUTOMATED COCHLEAR-IMPLANT INSERTION TECHNIQUE: Sushrut Kale, Mailing Wu, Elizabeth Olson
T22: AN ECAP DETECTION ALGORITHM BASED ON PRIOR SIGNAL KNOWLEDGE, SUITABLE FOR EMBEDDED COCHLEAR IMPLANT SIGNAL PROCESSING: Nicolas Veau, Jonathan Laudanski, Kai Dang, Yves Wenzinger, Jean-Marc Edeline, Boris Gourevitch
T23: STIMULATION OF THE GUINEA PIG COCHLEA WITH TRAINS OF AMPLITUDE MODULATED LASER PULSES: Claus-Peter Richter, Hunter Young, Xiaodong Tan
T24: DOES NEURAL SHUNTING LIMIT TEMPORAL CODING BY THE AUDITORY NERVE AT HIGH PULSE RATES? : David E. O'Gorman, Andy Brughera, H Steven Colburn
T25: WHY SO MANY NEURONS ARE NOT RESPONSIVE TO CI STIMULATION IN AUDITORY CORTEX?:Luke A. Johnson, Kai Y. Lim, Charles C. Della Santina and Xiaoqin Wang
T26: PREDICTING LOUDNESS GROWTH FROM THRESHOLDS MEASURED WITH BROAD AND FOCUSED ELECTRODE CONFIGURATIONS: Steven M Bierer, Joshua H Goldwyn, Julie Arenberg Bierer
T27: PRIMATE STUDIES OF VESTIBULAR IMPLANTATION: SINGLE NEURON, EVOKED POTENTIAL AND EYE MOVEMENT RECORDINGS: James Phillips, Leo Ling, Trey Oxford, Kaibao Nie, Amy Nowack, Sarah Shepherd, Christopher Phillips, Jay Rubinstein
T28: TRACKING BRAIN PLASTICITY WITH EVENT-RELATED OPTICAL SIGNAL (EROS) : Chin-Hong Tan, Michael A Novak, Chun-Yu Tse, Jennifer Black, Brian A Gordon, Ed Maclin, Benjamin Zimmerman, Gabriele Gratton, Monica Fabiani
T29: LONG DURATIONS OF UNILATERAL COCHLEAR IMPLANT USE DO NOT PROTECT THE UN-STIMULATED PATHWAYS FROM EFFECTS OF AUDITORY DEPRIVATION : Salima Jiwani, Blake C. Papsin, Karen A. Gordon
T30: EFFECTS OF DEAFNESS, CHRONIC INTRACOCHLEAR ELECTRIC STIMULATION AND BEHAVIORAL TRAINING ON NEURONAL TEMPORAL PROCESSING IN ICC AND AI: Maike Vollmer, Ralph E. Beitel, Russell L. Snyder, Patricia A. Leake
T31: PITCH PERCEPTION IN COCHLEAR IMPLANT USERS WITH SQUARE WAVE ITERATED RIPPLED NOISE: Richard T Penninger, Eugen Kludt, Marc Leman, Ingeborg Dhooge, Andreas Buechner
T32: ELECTRO-ACOUSTIC PITCH MATCHING IN BIMODAL COCHLEAR IMPLANT USERS: Mohammad Maarefvand, Peter Blamey, Julia Sarant, Jeremy Marozeau
T33: PERCEPTION OF COCHLEAR-IMPLANT SIMULATED SPEECH BETTER IN MUSICIANS: Christina Fuller, Rolien Free, John J. Galvin, QianJie Fu , Bert Maat, Deniz Baskent
T34: TRAINING TO IMPROVE PITCH PERCEPTION OF COCHLEAR IMPLANT USERS IN THE PRESENCE OF COMPETING PLACE CUES : Andrew Vandali, Richard van Hoesel, David Sly, Robert Cowan
T35: MELODIC PITCH PERCEPTION AND LEXICAL TONE PERCEPTION IN MANDARIN-SPEAKING COCHLEAR IMPLANT USERS: Duoduo Tao, Rui Deng, Qianjie Fu, John J Galvin III, Bing Chen
T36: EVALUATION OF MUSIC PRE-PROCESSING SCHEME WITH COCHLEAR IMPLANT RECIPIENTS: Wim Buyens[1 2 3], Bas van Dijk[1], Marc Moonen[2], Jan Wouters[3]
T37: THE EFFECTS OF SOUND CODING PARAMETERS ON THE PERCEPTION OF PROSODY BY COCHLEAR IMPLANT USERS: Chris James, Mathieu Marx, Olivier Deguine, Marie-Laurence Laborde, Bernard Fraysse
T38: HOW DOES ELECTRIC HEARING COMBINE WITH ACOUSTIC HEARING FOR SPEECH AND MUSIC?:Joseph Crew[1,2], John Galvin III[2], and Qian-Jie Fu[1,2]
T39: PITCH PERCEPTION AND PRODUCTION FOR HEARING-IMPAIRED CHILDREN WITH DIFFERENT HEARING DEVICES : Sian Louise Edwards, Merle H Mahon, Deborah A Vickers
T40: NEW MUSIC COMPOSITIONS FOR COCHLEAR IMPLANT USERS: Waldo Nogueira
T41: THE BENEFIT OF A UNILATERAL COCHLEAR IMPLANT IN PATIENTS WITH REGULAR HEARING OF THE CONTRALATERAL EAR: Mark Praetorius, Maria RoesliKhabas , Sebastian Hoth

T42: AN ALGORITHM TO ENHANCE THE BINAURAL CUES AVAILABLE TO BILATERAL COCHLEAR IMPLANT USERS: Christopher A. Brown
T43: THE BINAURAL INTERACTION COMPONENT OF THE CORTICAL AUDITORY EVOKED POTENTIAL IN CHILDREN: Shuman He, John H. Grose, Holly F.B. Teagle, Craig A. Buchman
T44: BILATERAL LOUDNESS BALANCING AND DISTORTED SPATIAL MAPS IN RECIPIENTS OF BILATERAL COCHLEAR IMPLANTS: Matthew B. Fitzgerald, Alan Kan, Matthew J. Goupell
T45: SELF-SELECTED AND PITCH-MATCHED FREQUENCY TABLES IN USERS OF BILATERAL COCHLEAR IMPLANTS: Matthew B. Fitzgerald, Katelyn Glassman, Keena Seward, Ksenia Prosolovich, Chin-Tuan Tan, Mario Svirsky
T46: LATERALIZATION AND BINAURAL FUSION FOR LONG INTERAURAL TIME DELAYS (ITDS) WITH BILATERAL COCHLEAR IMPLANTS: Victor A. Noel[1], Bertrand Delgutte[1,2,3] and Donald K. Eddington[1,2]
T47: PERCEPTUAL BENEFIT OF A LOW-FREQUENCY ACOUSTIC SIGNAL WITH ELECTRIC STIMULATION FOR CHILDREN AND ADULTS: Susan Nittrouer, Eric Tarr, Joanna Lowenstein
T48: BIMODAL COCHLEAR IMPLANTS: DO WE NEED TO BALANCE THE LEVEL OF THE ELECTRIC AND ACOUSTIC SIGNALS?: Michael F. Dorman, Philip Loizou, Shuai Wang, Ting Zhang, Tony Spahr, Louise Loiselle, Sarah Cook
T49: BINAURAL INTERACTION IN UNILATERAL COCHLEAR IMPLANT USERS WITH BILATERAL LOW-FREQUENCY RESIDUAL HEARING: Lindsey Van Yper, Katrien Vermeire, Ingeborg Dhooge
T50: BINAURAL SUMMATION OF SPECTRALLY SIDE-SEPARATED SPEECH AND BINAURAL FUSION OF TEMPORALLY ALTERNATING SPEECH IN COCHLEAR IMPLANT RECIPIENTS WITH UNILATERAL SEVERE TO PROFOUND HEARING LOSS: Thomas Wesarg, Susan Arndt, Antje Aschendorff, Natalie Zimmermann, Florian Langner, Karsten Plotz, Horst Hessel, Roland Laszig
T51: THE VIRTUAL REALITY AS A TOOL FOR THE INVESTIGATION OF THE LOCALIZATION WITH SINGLE SIDED DEAF COCHLEA IMPLANT USERS: Daniel Visser[1], Sven Schoernich[2,3], Joachim Mueller[1]
T52: FREQUENCY DEPENDENT LOUDNESS BALANCING TO OPTIMIZE HEARING AID GAINS IN BIMODAL CI USERS: Lidwien C. E. Veugen, Josef Chalupper, Ad F M Snik, John A. van Opstal, Lucas H. M. Mens
T53: HEAD MOVEMENT BENEFIT FOR SOUND LOCALIZATION IN SIMULATIONS OF COCHLEAR IMPLANT AND ELECTROACOUSTIC STIMULATION PAIRED WITH CONTRALATERAL HEARING AID, COCHLEAR IMPLANT, OR ELECTROACOUSTIC STIMULATION: Kassandra-Anne H. Birtch-Kaminskas[1]& Ewan A. Macpherson[1,2]
T54: IMPROVED AESTHETIC RESPONSES TO MUSIC IN BIMODAL AND BILATERAL COCHLEAR IMPLANT USERS: Hamish Innes-Brown, Nicolas Vannson, Isabelle Peretz, Jeremy Marozeau
T55: DO COCHLEAR IMPLANT USERS RELY ON DIFFERENT ACOUSTIC CUES THAN NORMAL-HEARING ADULTS? A STUDY OF CUE WEIGHTING AND SALIENCE: Aaron C Moberly, Joanna Lowenstein, Eric Tarr, Amanda Caldwell, D Bradley Welling, Antoine Shahin, Susan Nittrouer
T56: A NEW CLINICAL SPEECH PERCEPTION TEST FOR SAUDI CHILDREN WITH COCHLEAR IMPLANTS: Nada Abdulkader Alsari, Andrew Faulkner
T57: IDENTIFICATION OF VOWELS IN /DVD/ SYLLABLES BY COCHLEAR IMPLANT USERS: RELIANCE ON VOWEL DURATION CUES: Gail S. Donaldson, Lindsay B. Johnson, Soo Hee Oh, Catherine L. Rogers
T58: THE EFFECT OF MICROPHONE LOCATION ON SPEECH INTELLIGIBILITY AND LOCALIZATION IN UNILATERAL CI USERS: Stefan Fredelake, Gunnar Geissler, Martina Brendel, Carolin Frohne-Buechner, Andreas Buechner, Thomas Lenarz

POSTER SESSION W: WEDNESDAY –8AM TO MIDNIGHT

W1: EFFECT OF MASKER LEVEL, RATE, AND SPATIAL SEPARATION ON PSYCHOPHYSICAL THRESHOLDS FOR INTERLEAVED PULSE TRAINS: Michelle L. Hughes, Kirsten Euscher, Jenny Goehring, Jacquelyn Baudhuin, Donna Neff

W2: ADAPTIVE ELECTRODE SELECTION USING PSYCHOPHYSICAL MEASURES: Sara I Duran, Leslie M Collins, Chandra S Throckmorton

W3: THE ROLE OF AUDITORY FEEDBACK ON VOCAL PITCH PRODUCTION IN COCHLEAR IMPLANT USERS: Juan Huang, Yang Zhang, Patpong Jiradejvong, Courtney Carver, Xiaoqin Wang, Charles Limb

W4: TEMPORAL PATTERN PERCEPTION USING MONOPOLAR AND TRIPOLAR STIMULATION: Claire Fielden, Karolina Kluk, Colette McKay

W5: LINKING TEMPORAL MASKING AND SPEECH UNDERSTANDING IN HEARING WITH COCHLEAR IMPLANTS: Bernhard U. Seeber, Robert H. Pierzycki

W6: EQUIVALENT MASKING PATTERNS OF BIPOLAR PROBES FORWARD-MASKED BY MONOPOLAR OR BIPOLAR MASKERS: Mohamed Bingabr, Blas Espinoza-Varas, Robert Smith, Saroj Sigdel

W7: IMPROVING PLACE-PITCH IN THE APEX VIA MANIPULATIONS IN PHASE DURATION: Joshua Stohl[1,2], Robert Wolford[1], Blake Wilson[2,3]

W8: UNDERSTANDING THE EFFECTS OF CENTRAL AND PERIPHERAL MASKING IN COCHLEAR IMPLANT USERS: Carol Q. Pham, Peter Bremen, Weidong Shen, Shiming Yang, John C. Middlebrooks, Fan Gang Zeng, Myles Mc Laughlin

W9: ACROSS-ELECTRODE PROCESSING IN COCHLEAR IMPLANT USERS: Stefan Dieter Zirn, John Martin Hempel, Maria Elke Schuster, Werner Hemmert

W10: SINGLE- AND MULTI-CHANNEL MODULATION DETECTION BY COCHLEAR IMPLANT USERS: John J. Galvin III,[1,3]Qian-Jie Fu,[1] Sandy Oba,[1] and Deniz Başkent [2,3]

W11: EXTRACTING INTERAURAL TEMPORAL INFORMATION FROM THE ENVELOPES OF LOW- AND HIGH-FREQUENCY MODULATED SOUNDS: Mathias Dietz, Torsten Marquardt, Andrew Simpson, David McAlpine

W12: PLACE AND RATE DEPENDENCY OF CHANNEL INTERACTION WITH SIMULTANEOUS STIMULATION: Reinhold Schatzer, Paul Bader, Andreas Griessner, Clemens Zierhofer

W13: COMPARISON OF TEMPORAL INTEGRATION AND MULTIPULSE INTEGRATION FUNCTIONS IN HUMANS WITH COCHLEAR IMPLANTS: Ning Zhou, Bryan E. Pfingst

W14: ASSESSING THE HEALTH OF THE IMPLANTED COCHLEA WITH NONINVASIVE MEASURES: Deborah J. Colesa, Gina L. Su, Melissa M. Watts, Yehoash Raphael, Bryan E. Pfingst

W15: WHAT DOES SPECTRAL RIPPLE DISCRIMINATION AT HIGH RIPPLE DENSITIES MEAN IN COCHLEAR IMPLANT LISTENING EXPERIMENTS?: Naomi B. Croghan, Harish Krishnamoorthi, Zachary M. Smith

W16: MINIMUM AUDIBLE MOVEMENT ANGLE IN BILATERAL COCHLEAR IMPLANT USERS: Wiebke Heeren, Michael Ziese, Jesko Verhey

W17: EFFECTS OF STIMULATION MODE ON TEMPORAL PROCESSING BY COCHLEAR IMPLANT LISTENERS.: Aditya M Kulkarni, Monita Chatterjee

W18: AUDITORY CONTEXT EFFECTS IN COCHLEAR-IMPLANT USERS: Ningyuan Wang, Heather Kreft, Andrew J. Oxenham

W19: SPREAD OF EXCITATION USING A NEW STIMULATION MODE, THE VIRTUAL TRIPOLE: Monica Padilla, David M Landsberger

W20: ELECTRODE SPANNING WITH PARTIAL TRIPOLAR STIMULATION MODE IN COCHLEAR IMPLANTS: Ching-Chih Wu and Xin Luo

W21: COMPARING MONOPOLAR WITH MULTIPOLAR STIMULATION MODES : Jeremy Marozeau, Hugh J McDermott, Brett A Swanson, Colette M McKay

W22: ASSESSING THE EFFECTS OF PULSE POLARITY ON SEQUENTIAL INTERACTIONS: Olivier Macherey, Robert P. Carlyon, Stephane Roman

W23: EFFECT OF PULSE POLARITY IN BIPHASIC AND TRIPHASIC PULSES : Sonja Andrea Karg, Marek Rudnicki, Christina Lackner, Werner Hemmert
W24: ECAP-BASED MEASUREMENTS OF FORWARD MASKING FUNCTIONS : Wai Kong Lai, Norbert Dillier
W25: EFFECT OF PULSE RATE ON THE DETECTION OF ELECTRICAL PULSE TRAINS BY USERS OF COCHLEAR AND AUDITORY BRAINSTEM IMPLANTS: Robert P Carlyon, John M Deeks, Colette M McKay
W26: EFFECT OF MEDIAL-LATERAL ELECTRODE DISTANCE ON PSYCHOPHYSICAL AND OBJECTIVE MEASURES: Christopher J Long, Tim A Holden, Zachary M Smith, Wendy S Parkinson
W27: RATE DISCRIMINATION AT LOW PULSE RATES: COMPARISON BETWEEN NORMAL-HEARING AND COCHLEAR IMPLANT LISTENERS AND INFLUENCE OF INTRACOCHLEAR STIMULATION SITE: Pierre Stahl, Olivier
W28: USING ELECTRODE DISCRIMINATION TO ENHANCE CLINICAL FITTING OF COCHLEAR IMPLANTS : Shaza M Saleh, Shakeel R Saeed, Leah Meerton, Debi A Vickers
W29: COCHLEAR IMPLANT LISTENERS DEMONSTRATE ELEVATION PERCEPTION FOR SINGLE ELECTRODE STIMULI: Tanvi D. Thakkar, Olga Stakhovskaya, Matthew J. Goupell
W30: SPECTRAL RIPPLE TESTING MODULATION DETECTION: AND EFFECT OF STARTING PHASE: Melanie Gilbert, Leonid M Litvak, Aniket Saoji
W32: EFFECT OF MASKER FLUCTUATIONS ON LEXICAL SEGMENTATION IN COCHLEAR IMPLANT LISTENERS: Trevor T Perry, Bomjun J Kwon
W33: APICAL VERSUS NON-APICAL ELECTRIC STIMULATION OF THE COCHLEA: Margaret T Dillon[1], Marcia C Adunka[2], English R King[2], Harold C Pillsbury[1], Oliver F Adunka[1], Craig A Buchman[1]
W34: THE BENEFIT OF HEAD ORIENTATION TO INTELLIGIBILITY OF SPEECH IN NOISE: Jacques Grange, John Culling
W35: CLINICAL FACTORS AFFECTING PERFORMANCE OF POSTLINGUISTICALLY DEAF ADULTS USING COCHLEAR IMPLANTS: A NEW CONCEPTUAL MODEL OVER TIME BASED ON 2251 PATIENTS : Diane S Lazard, Peter J Blamey
W36: EFFECTS OF NUMBER OF ELECTRODES ON LISTENING EFFORT IN COCHLEAR IMPLANT USERS: Carina Pals, Anastasios Sarampalis, Deniz Başkent
W37: IMPACT OF REVERBERATION ON SPEECH PERCEPTION WITH COCHLEAR IMPLANTS: Uwe Baumann, Hannah-Lina Grahlmann, Tobias Rader, Matthias Milczynski, Tobias Weissgerber
W38: SIMULATING THE BIMODAL EXCITATION SPREAD PRODUCED BY BIPOLAR STIMULATION: EFFECT ON SPEECH INTELLIGIBILITY : Quentin Mesnildrey, Olivier Macherey
W39: USE OF PSYCHOACOUSTIC MEASURES TO EVALUATE COCHLEAR IMPLANT CANDIDACY: Il Joon Moon, Hyun Joon Shim, Jong Ho Won, Ward R Drennan, Elizabeth S Anderson, Jay T Rubinstein
W40: ROLE OF BINAURAL TEMPORAL FINE STRUCTURE AND ENVELOPE CUES FOR SPATIAL RELEASE FROM MASKING: Jayaganesh Swaminathan, Christine R Mason, Timoth M Streeter, Virginia Best, Gerald Kidd Jr
W41: MEASURING SPECTRAL RESOLUTION FOR SPEECH SOUNDS: IMPLICATIONS FOR COCHLEAR IMPLANTS AND NEW CODING STRATEGIES: Matthew Winn, Ruth Litovsky
W42: COMPARISON OF VOWEL CONFUSIONS FROM VOCODER SIMULATIONS IN NORMAL HEARING AND COCHLEAR IMPLANT LISTENERS: Mishaela DiNino, Kelley C. Corcoran, Julie Arenberg Bierer
W43: PSYCHOMETRIC FUNCTION OF RATE PITCH DISCRIMINATION DEPENDING ON JITTER APPLIED ON THE RATE OF ELECTRICAL PULSE PATTERN.: Andreas Bahmer, Uwe Baumann
W44: CLINICAL ASSESSMENT OF SPECTRAL MODULATION DETECTION: Rene Headrick Gifford, Anthony J Spahr, Andrea Hedley-Williams
W45: VALIDATION OF A CLINICAL ASSESSMENT OF SPECTRAL RIPPLE DISCRIMINATION ABILITY: Ward R. Drennan[1], Elizabeth S. Anderson[1], Jong Ho Won[2] and Jay T. Rubinstein[1]

W46: EFFECT OF ELECTRODE IMPEDANCE ON ECAP SPREAD OF EXCITATION AND PITCH DISCRIMINATION WITH DUAL-ELECTRODE STIMULATION: Michelle L. Hughes, Jacquelyn L. Baudhuin, Jenny L. Goehring, Donna L. Neff

W47: DETERMINING THE BOUNDARY CONDITIONS FOR COCHLEAR IMPLANT MODELS: Phillip Tran, Paul Wong, Andrian Sue, Qing Li, Paul Carter

W48: AN INFORMATION THEORETIC FRAMEWORK FOR THE EVALUATION OF COCHLEAR IMPLANT CODING STRATEGIES: Jonathan Laudanski, Dan Gnansia, Teng Huang, Nicolas Veau

W49: DIMENSIONAL ANALYSIS OF THE HUMAN COCHLEA: Ersin Avci, Tim Nauwelaers, Thomas Lenarz, Volkmar Hamacher, Andrej Kral

W50: A NEW GRAPHICAL METHOD FOR SHOWING THE ELECTRICALLY STIMULATED HUMAN COCHLEA: Mark E Robert, Frederick H Linthicum

W51: OPTICAL STIMULATION OF THE COCHLEA - SINGLE CELL MEASUREMENTS ON HEK-HCLC CELLS AND SPIRAL GANGLION NEURONS: Alexander Rettenmaier, Athanasia Warnecke, Thomas Lenarz, Guenter Reuter

W52: INFLUENCE OF “DEAD” REGIONS ON THE TONOTOPIC REPRESENTATION OF SPEECH: Michele Nicoletti , Werner Hemmert

W53: A PHENOMENOLOGICAL MODEL OF THE AUDITORY NERVE FIBER RESPONSE TO MONOPHASIC AND BIPHASIC ELECTRICAL STIMULATION: Colin DF Horne, Chris J Sumner, Bernhard U Seeber

W54: MODEL-BASED EVALUATION OF SPEECH PROCESSING STRATEGIES IN A PITCH DISCRIMINATION TASK: Nafise Nina Erfanian Saeedi[1,2], Peter Blamey[3], Anthony N. Burkitt[1,2,3], David B. Grayden[1,2,3]

W55: MODELING AND EVALUATION OF LARGE SPIRAL GANGLION POPULATION RESPONSES: Michele Nicoletti, Michael Isik, Christian Wirtz, Werner Hemmert

W56: CORRELATION AND TEMPORAL PROPERTIES OF THE AUDITORY NERVE: Brian Dougherty, Claus-Peter Richter

W57: MYOGRAPHIC RECORDING OF THE ELECTRICALLY ELICITED STAPEDIUS REFLEX: Rolf Battmer, Dietmar Basta, Ingo Todt, Arneborg Ernst

W58: ELECTROCOCHLEOGRAPHY ELICITED BY AN ELECTRICAL STIMULATION ON THE PROMONTORY: Giacomo Mandruzzato, Marek Polak

W59: AUDITORY EVOKED POTENTIALS WITH DIRECT ACOUSTIC COCHLEAR STIMULATION: Nicolas Verhaert, Michael Hofmann, Jan Wouters

W60: EFFECT OF TRIPHASIC AND BIPHASIC PULSE SHAPES ON THE SPREAD OF EXCITATION David Bakhos, Griet Mertens, Paul Van De Heyning, David M. Landsberger

W61: OF NOISE-SUPPRESSED SPEECH BY COCHLEAR IMPLANT USERS: Fei Chen, Tian Guan, Lena Wong, Yi Hu

POSTER SESSION R: THURSDAY –8AM TO MIDNIGHT

R1: PULSE-SPREADING HARMONIC COMPLEXES: AN ADVANTAGEOUS CARRIER FOR SIMULATING COCHLEAR IMPLANTS: Gaston Hilkhuysen, Quentin Mesnildrey, Olivier Macherey

R2: AMPLITUDE MODULATION DETECTION AND SPEECH RECOGNITION IN LATE IMPLANTED PRELINGUALLY DEAFENED COCHLEAR IMPLANT USERS: Anke M. De Ruiter(1,2), Joke A. Debruyne(1,3), Michelene N. Chenault(1), Jan P.L. Brokx(1,3)

R3: PERIODICITY AND APERIODICITY IN THE PERCEPTION OF SPEECH IN BOTH STEADY-STATE AND FLUCTUATING MASKERS: Stuart Rosen, Tim Green, Marine Ardoint , Andrew Faulkner

R4: LITTLE RED RIDING HOOD WAS A COCHLEAR IMPLANT USER!! Etienne Gaudrain, Christina Fuller}*, Jeanne Clarke, John J Galvin, Qian-Jie Fu, Rolien Free, Deniz Başkent

R5: TRADING FUNDAMENTAL FREQUENCY FOR DURATION CUES IN LEXICAL TONE RECOGNITION BY TAIWANESE CHILDREN WITH COCHLEAR IMPLANTS: Shu-Chen Peng, Hui-Ping Lu, Mickael LD Deroche, Nelson Lu, Yung-Song Lin, Monita Chatterjee

R6: EVALUATION OF A REAL-TIME IMPLEMENTATION OF AN EXPLICIT PITCH CODING STRATEGY FOR COCHLEAR IMPLANTS: Alejandro Osses, Jan Wouters, Tom Francart

R7: RELEASE FROM MASKING OF SPEECH IN 8 HZ SAM NOISE IN THE ABSENCE OF TEMPORAL FINE STRUCTURE: Enja Jung, Andrew Faulkner, Stuart Rosen, Tim Green

R8: RESULTS IN ADULT COCHLEAR IMPLANT USERS WITH ASYMMETRIC HEARING LOSS: Jill B. Firszt, Ruth M. Reeder, Laura K. Holden, Sarah Zlomke, Kristen Lewis

R9: ACOUSTIC SPECTRO-TEMPORAL MODULATION DETECTION IN COCHLEAR IMPLANT USERS: Corey Stoelb, Yi Zheng, Monty Escabi, Ruth Litovsky

R10: INFLUENCE OF EXPANDED VOICE PITCH CONTOURS ON THE INTELLIGIBILITY OF SPECTRALLY REDUCED SPEECH: Christopher A. Brown, Kate Helms-Tillery

R11: TEMPORAL RESPONSE PROPERTIES OF THE AUDITORY NERVE IN CHILDREN WITH AUDITORY NEUROPATHY SPECTRUM DISORDER: Shuman He, John H. Grose, Holly F.B. Teagle, Craig A. Buchman

R12: EFFECT OF CONSONANT-VOWEL BOUNDARY TO SPEECH PERCEPTION IN COCHLEAR IMPLANTS: Nirmal Kumar Srinivasan, Hussnain Ali, Emily Tobey, Philipos C Loizou

R13: IS WEAK PITCH FULLY RESPONSIBLE FOR LOW COMPREHENSION OF INTERRUPTED SPEECH WITH POOR SPECTRAL RESOLUTION?: Jeanne Clarke, Etienne Gaudrain, Deniz Baskent

R14: PRESTO: PRELIMINARY FINDINGS WITH A NEW HIGH-VARIABILITY SENTENCE RECOGNITION TEST IN PATIENTS WITH COCHLEAR IMPLANTS: Kathleen F Faulkner, David B Pisoni

R15: EFFECT OF INHERENT AMPLITUDE FLUCTUATIONS IN STATIONARY NOISE ON SPEECH INTELLIGIBILITY IN COCHLEAR-IMPLANT USERS: Heather Kreft, Andrew J. Oxenham

R16: ASSESSING LISTENING EFFORT IN NORMAL-HEARING AND COCHLEAR IMPLANT RECIPIENTS THROUGH USE OF A DUAL-TASK PARADIGM: Matthew B. Fitzgerald, Keena Seward, Katelyn Glassman, Nandini Govil, Arlene C. Neuman

R17: EXPLORING MASKING RELEASE FOR DIFFERENT MASKER MODULATION RATES USING A NEAR-STEADY MASKER : Michael A Stone, Brian CJ Moore

R18: A TWO-MICROPHONE SPEECH ENHANCEMENT ALGORITHM DEDICATED TO A BILATERAL COCHLEAR IMPLANT: Fathi Kallel, Christian Berger Vachon, Ahmed Ben Hamida

R19: A PITCH SYNCHRONOUS FRAMEWORK FOR REPRESENTING, RECOVERING, AND MODIFYING TEMPORAL STRUCTURE IN SPEECH: Eric Tarr, Ashok Krishnamurthy, Susan Nittrouer

R20: A SOFT MASKING STRATEGY FOR SIMULTANEOUS SUPPRESSION OF NOISE AND REVERBERATION IN COCHLEAR IMPLANTS: Oldooz Hazrati, Seyed Omid Sadjadi, Hussnain Ali, Philipos C. Loizou, John H. L. Hansen

R21: ENHANCING RAPID VARIATIONS OF COCHLEAR IMPLANT SIGNAL USING A SIMPLE MODEL OF SHORT-TERM ADAPTATION: Mahan Azadpour, Robert L. Smith
R22: RELATIVE STRENGTH OF PLACE PITCH CUES RESULTING FROM CURRENT STEERING AND FROM ELECTRODE TRANSITION EFFECTS: Johan Hanekom, Pieter Venter
R23: PARTIAL TRIPOLAR STIMULATION IMPROVES SPEECH IN NOISE AND SPECTRAL RESOLUTION: David M Landsberger, Arthi G Srinivasan, Justin M Aronoff, Bonnie Ong, Joe Crew
R24: PROGRESS AND CHALLENGES IN HUMAN VESTIBULAR IMPLANTATION: Jay T Rubinstein, James O Phillips, Christopher Phillips, Kaibao Nie, Leo Ling, Amy Nowack, Christina DeFrancesci
R25: IMPROVED PERCEPTION OF MUSIC AND MANDARIN TONES WITH A HARMONIC CODING STRATEGY FOR COCHLEAR IMPLANTS: Kaibao Nie, Xing Li, Les Atlas, Ward Drennan, Jay Rubinstein
R26: ACES: A NEW SPEECH CODING STRATEGY FOR COCHLEAR IMPLANTS WHICH COMBINES THE BENEFITS OF CIS AND ACE CODING STRATEGIES: Obaid ur Rehman Qazi[1,2,3], Bas van Dijk[1], Marc Moonen[2] and Jan Wouters[3]
R27: PITCH DISCRIMINATION OF AMPLITUDE-MODULATED CURRENT PULSE TRAINS WITH HARMONIC RELATED CARRIER PULSE RATE.: Eugen Kludt, Thomas Lenarz, Andreas Buechner
R28: AN OBJECTIVE EVALUATION OF THE DE-REVERBERATION EFFECT OF COCHLEAR-IMPLANT AND HEARING AID FRONTEND ALGORITHMS: Matthias Milczynski, Karl-Heinz Dyballa, Andreas Buechner, Phillipp Hehrmann, Martin Kuster, Volkmar Hamacher
R29: COCHLEAR IMPLANT NOISE REDUCTION: PERFORMANCE, QUALITY AND ACCEPTANCE: Stefan J Mauger, Komal Arora, Pam W Dawson, Adam A Hersbach, John M Heasman
R30: TWO NOVEL CODING STRATEGIES BASED ON NEURAL EXCITABILITY: Wai Kong Lai, Matthijs Killian, Onur Babacan, Norbert Dillier
R31: IMPROVING COCHLEAR IMPLANT OUTCOME THROUGH ALTERED AUTOMATIC GAIN CONTROL AND COMPRESSION SETTINGS: Patrick J Boyle, Michael A Stone, Brian C.J. Moore
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SPEAKER ABSTRACTS

S1: COCHLEAR IMPLANTS AND TONAL LANGUAGES

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Poor representation of pitch information in cochlear implant(CI) deviceshinders pitch perception and affects perception of lexicaltones in cochlear implant users who speak tonal languages [e.g., Wang et al., 2011, *Int. J. Audiol*., 50, 270-278].

One hundred ten Mandarin-speaking, prelingually-deafened CI subjects (age: 2.5 to 16 years) and 125 typically developing, normal-hearingsubjects (age: 3 to 10 years) were recruited from Beijing and Shanghai, China. Lexicaltone perception in quiet and in noise (at +12, +6 and 0 dB signal-to-noise ratios) was measured using a computerized tone contrasttest. Tone production was judged by native Mandarin-speakingadult listeners as well as analyzed acoustically and with an artificialneural network.Ageneral linear model analysiswas performedto determine factors that accounted for performance variability.

Prelingually-deafened children with CIs scored from chance level to nearly perfect performance on the lexical toneperception task. Moderate amount of noise had very small effects on tone perception in normal-hearing children but had tremendous effects on tone perception in children with CIs. Brand of CI device (i.e., Advanced Bionics, Cochlear, or MedEl) did not show significant effects on tone perception performance. The degree of differentiation of tones producedby the CI group was significantly lower than the control group asrevealed by acoustic analysis. Tone production performanceassessed by the neural network was highly correlated with thatevaluated by human listeners (r= 0.94). There was a moderate correlationbetween the overall tone perception in quiet and tone production performanceacross CI subjects (r=0.56). Duration of implant use and age at implantationjointly explained approximately 30% of the variance in the tone perceptionperformance. Age at implantation was the only significant predictorfor tone production performance in children with CIs.

While accurate representation of lexical tones may not be essential for tonal language sentence recognition in quiet [Feng et al., 2012, *J. Acoust. Soc. Am*., 131, EL133-EL138], it is particularly important for sentence recognition in noise [Chen et al., 2013]. Children with CIs demonstratesuboptimal tone perception in quiet and yet very poor tone perception in noise. Tone production performance in pediatric CI usersis dependent on accurate perception.Early implantation predictsa better outcome in lexical tone perception and production.

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S2: INTRINSIC AND EXTRINSIC PITCH CUES TO TONE RECOGNITION WITH COCHLEAR IMPLANTS

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Mandarin tones (Tone 1: high-level, Tone 2: mid-rising, Tone 3: low-falling-rising, and Tone 4: high-falling) are characterized by the different pitch levels and contours of target vowels and are used to contrast word meanings. Given the limited access to pitch cues, cochlear implant (CI) users only have moderate levels of tone recognition by using secondary cues such as vowel duration and amplitude envelope in addition to temporal periodicity cues. In this talk, I would like to review our recent studies on both intrinsic pitch cues within the target tones and extrinsic pitch cues in the context that may contribute to tone recognition with CIs. These results will shed lights on the tone recognition mechanisms and suggest ways to enhance tone recognition in CI users.

The first study tested two types of pitch cues to tone recognition, based on CI place and temporal coding, respectively. Current steering between adjacent electrodes and amplitude modulation (AM) frequency changes were separately or simultaneously used to encode rising and falling pitch contours, simulating Mandarin Tones 2 and 4. When the time-varying place and temporal pitch cues were similarly discriminable and consistent with each other, CI users integrated them to achieve better pitch contour identification. These results suggest that tone recognition with CIs may be enhanced by coordinating current steering and AM frequency changes within the target tones.

The second study tested the talker variability effect on tone recognition in normal hearing (NH) listeners listening to acoustic CI simulations with different spectral and temporal cues. The tone stimuli within each test block were randomly selected from a single talker in the single-talker condition or from four different talkers in the multi-talker condition. Unlike previous NH results, the difference in tone recognition between the single- and multi-talker conditions was inconsistent across subjects and CI simulations. These results suggest that CI users may not use the talker-specific pitch cues from previous test tokens in the single-talker condition to better recognize target tones.

The third study tested the effect of context pitch on tone recognition with CIs. Previous studies have shown that Chinese NH listeners identified the same target tones more often as Tone 2 (mid-rising) following a high-f0 sentence but as Tone 1 (high-level) more often following a low-f0 sentence. The context effect may help NH listeners resolve the talker variability in voice pitch for tone recognition. Similar context effect was found for Chinese NH listeners, while weaker context effect was found for English NH listeners on the identification of non-speech pitch contours simulating Mandarin Tones 1 and 2. Thus, the context effect involves general auditory processing of pitch cues and strengthens with tonal language experience. Interestingly, English CI users also had a contrastive context effect on pitch contour identification, despite their degraded peripheral inputs. Taken together, Chinese CI users are expected to make even stronger contrastive use of context pitch cues in tone recognition as a way of talker normalization. This hypothesis is currently being tested in Chinese CI users.

Research was supported by NIH/NIDCD grant R21-DC011844 and Purdue University.

S3: PROCESSING OF PITCH, PROSODY, VOICE EMOTION, AND LEXICAL TONES BY COCHLEAR IMPLANTED CHILDREN

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Fundamental frequency (F0, or voice pitch) coding remains a problem in present-day cochlear implants (CIs). While transmission of musical pitch via CIs remains a distant goal, prosodic and lexical information may demand less fine-grained pitch resolution. The ability to recognize prosodic cues is an important first step for spoken language acquisition in young children. For children speaking tonal languages, lexical tone recognition is also an important aspect for linguistic comprehension. Finally, emotion recognition is an important aspect of children's communicative and social development. In all of these cases, voice pitch inflections are the dominant acoustic cue. However, other co-varying cues, such as intensity and duration changes, may be used in the absence of a salient pitch cue. We are investigating how children with CIs (i) discriminate F0 cues, (ii) use F0 vs. other co-varying cues in a question-statement identification task, (iii) use F0 vs. other co-varying cues in lexical tone recognition, (iv) perform in voice emotion recognition tasks, and (v) compare with their normally hearing (NH) peers in all these tasks.

To date, our results indicate that (i) CI children have poorer F0 discrimination abilities than their NH peers; (ii) NH children rely mostly on F0 cues, ignoring intensity and duration cues, for speech intonation recognition, while CI children rely less of F0 cues and more on the other cues in the same task; (iii) Mandarin-speaking, NH children rely primarily on F0 cues for lexical tone recognition, while CI Mandarin-speaking, CI children rely less on F0 cues and more on the secondary cue of duration; (v) CI children show poorer performance in voice emotion recognition than their NH peers.

These results suggest that, despite their greater difficulty with F0 discrimination, children with CIs can compensate by using co-varying acoustic cues in everyday communication. Ongoing analyses are investigating the effects of age of implantation, chronological age, and linguistic background on their performance. These results have important implications for aural rehabilitation of children with cochlear implants.

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S4: LIMITATIONS OF TEMPORAL PITCH PERCEPTION MEASURED USING THE AUDITORY NERVE RESPONSE TO ELECTRICAL STIMULATION

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A goal of several new cochlear implant (CI) speech-processing strategies is to improve the ability of patients to extract a pitch from the temporal pattern of electrical stimulation. Even with idealized stimuli, however, the ability of CI users to extract this temporal pitch is limited. We investigated the neural basis of two phenomena that reflect limitations on temporal pitch perception: (i) detection of changes in the rate of isochronous pulse trains usually deteriorates for baseline rates higher than 300 pps (the "upper limit" of temporal pitch), and (ii) when pulse trains are not perfectly isochronous – for example with inter-pulse intervals that alternate between 4 and 6 ms – perceived pitch varies markedly across listeners and corresponds to an inter-pulse interval (IPI) longer than the mean IPI of the stimulus (e.g. 5.7 ms for "4-6" pulse train). Both phenomena might be due to the compound neural response – as measured by the ECAP – alternating between a high and a low value from pulse to pulse. This could cause intervals equal to that between every other pulse to be conveyed to the brain, thereby lowering pitch. To test this we measured ECAPs and behavioural responses to the same stimuli (pulse trains) and with the same subjects. Results obtained with 4-6 and 3-7 pulse trains were qualitatively consistent with auditory nerve (AN) refractory effects influencing pitch: ECAP modulation was greater, and perceived pitch lower, for the 3-7 than for the 4-6 pulse train. For isochronous pulse trains, ECAP modulation increased and rate discrimination deteriorated with increasing rate, again qualitatively consistent with a role for AN refractoriness. However, a control experiment, in which we physically reduced the amplitude of alternate pulses in a 200-pps train, revealed that substantial ECAP modulations could correspond to only a small effect on pitch. Overall, the size of the ECAP modulations produced either by the alternating-interval or isochronous stimuli were not large enough to produce the observed perceptual effects. This suggests that temporal pitch is limited either by processes central to the AN, or by aspects of the temporal response not measured by averaging the ECAP over multiple sweeps. We are testing this latter hypothesis by investigating the statistics of the ECAP response measured on a sweep-by-sweep basis.

S5: FACILITATION, ACCOMMODATION AND REFRACTION IN ELECTRICALLY-STIMULATED MODEL AUDITORY NERVE FIBERS IS DEPENDENT ON LOCATION OF STIMULATION AND VOLTAGE-GATED CHANNEL TYPE DISTRIBUTION

Jason Boulet, Ian Bruce
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Voltage-gated ion channels play an important role in regulating the response characteristics of auditory nerve fibers (ANFs). A number of electrophysiological studies of ANFs stimulated by cochlear implants have demonstrated a range of temporal interactions for pulsatile stimulation including refraction, spike-rate adaptation, facilitation, and accommodation. The latter two phenomena refer to mechanisms by which previous sub-threshold stimulation can increase or decrease, respectively, the excitability to the next stimulus. A computational model of an ANF node of Ranvier (Negm& Bruce, EMBC 2008) suggests that low-threshold potassium (KLT) and hyperpolarization-activated cyclic nucleotide-gated cation (HCN) channels affect an ANF's absolute and relative refractory periods and produce adaptation/accommodation to high-rate pulse trains. However, for ongoing pulse trains it is difficult to delineate the individual contributions of these different forms of temporal interaction.

In this study, we addressed this issue by using a compartmental model of an entire ANF and applying the paired pulse paradigm of Cartee et al. (Hearing Res 2000, 2006) to examine facilitation, accommodation and refraction as a function of 1) the location of electrical stimulation and 2) nodal channel composition at important locations along the ANF. We tested three ANF models variants: A) only fast Nav and delayed-rectifier Kv at all nodes, B) with the addition of KLT & HCN channels at the first peripheral node and on the nodes of Ranvier neighboring the soma and C) by expanding the distribution of KLT channels to all nodes.

When using a paired pulse paradigm with a maximum inter-pulse interval of 0.3 ms, in cases where the first pulse was sub-threshold Cartee and colleagues only observed facilitation. By extending the range of inter-pulse intervals in our simulations, we were able to observe a combination of facilitation and accommodation for all models and at all nodes, although accommodation tended to dominate at nodes and neighboring nodes where KLT & HCN channels were located. Facilitation preceded accommodation, but there was often substantial overlap in their time-courses. In cases where the first pulse was supra-threshold, refraction was observed in the response to the second pulse. The absolute refractory period followed a similar pattern to the KLT & HCN location-dependent accommodation by being larger in those areas.

In conclusion, we argue that the combined effects of refraction, facilitation and accommodation bear a significant portion of the responsibility for explaining the transient and steady-state spike rates observed in physiological studies of ANF responses to high-rate pulsatile electrical stimulation.

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S6: REDUCING ELECTRICAL INTERACTION DURING PARALLEL STIMULATION USING VARIOUS COMPENSATION TECHNIQUES

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To improve spatial selectivity, various multipolar stimulation paradigms were tested in recent years. In general more charge is required to reach sufficient loudness, in turn requiring broader pulses and reduced rates to stay within voltage compliance limits. The use of parallel stimulation is a possible solution for this problem. However, this was associated with performance degradation in the past, most likely as a consequence of electrical interaction. This study uses psychophysics and computational modeling to evaluate three methods to reduce electrical interaction for parallel stimulation.
The compensation methods used targeted to get the same neural excitation elicited by sequential monopolar stimulation and were compared with standard simultaneous stimulation. The following compensation methods were evaluated: the Channel Interaction Compensation (CIC) algorithm (Zierhofer, 2008), partial tripolar stimulation (pTP) and a phased array compensation (PAC) method. In this last method the target was not to achieve a zero potential along the array but to get a field equal to the superposition of two monopolar current sources.

In both model and patients (n=8) the loudness growth was evaluated for all four situations, where ideally the same growth of the sequential monopoles would be found. Additionally, the interaction between simultaneously stimulated maskers (with 3 or 5 contacts in between) was measured in the patients using forward masking experiments, with a monopolar probe contact exactly in between the two maskers. The model allowed for detailed analysis of the neural excitation patterns created by the different stimuli on a range of levels.

Preliminary results indicate that with just 3 contacts between the maskers, none of the compensation methods is successful in reducing the threshold shifts caused by interaction between them. For the wider separation detection thresholds are all slightly lower, but only the PAC is in the same range as the sequential stimulus. The loudness experiments show that CIC and the PAC restore the loudness growth towards that of the sequential paradigm while pTP needs much more current.

The preliminary conclusion is that it is possible to compensate for the negative effects of electrical interaction during parallel stimulation at least at the level of excitation patterns. Future research has to demonstrate whether this will also lead to better speech perception.

Support provided by Advanced Bionics and the Heinsius-Houbolt fund.

S7: SOME OBSERVATIONS ON THE INDEXICAL PROPERTIES OF SPEECH: WHAT ARE THEY AND WHY DO WE NEED THEM FOR ROBUST SPEECH RECOGNITION AND SPOKEN LANGUAGE UNDERSTANDING?

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Speech is a "special" biologically-significant acoustic signal that simultaneously encodes two parallel channels or streams of information generated by a human talker engaged in a communicative act. On the one hand, the complex time-varying speech waveform carries the talker's intended linguistic message—the words and sentences of a natural language. On the other hand, the speech waveform also encodes and transmits reliable information about the vocal source—the talker's physical, social and mental states. This complementary channel of the speech waveform provides important contextual information about the talker's gender, regional dialect, emotional, and physical condition. Normal-hearing listeners automatically and unconsciously encode, store and process vocal source information along with the symbolic content of the message. These two parallel channels of speech are inseparable in both speech production and perception although the "duplex" nature of speech is not widely recognized or appreciated by speech and hearing scientists, especially audiologists and auditory neuroscientists working with hearing impaired populations. Everyone knows about the linguistic properties of speech—the well-known "speech cues" that underlie speech intelligibility and speech recognition. But what about the indexical properties of speech? What are they? What do these attributes of the speech signal do for the listener? What is their function in speech communication and why do we need them to understand speech recognition and spoken language, especially in hearing-impaired populations? In this talk, I offer some observations about the duplex nature of the speech signal. I argue that speech is not a linear sequence of idealized abstract context-free linguistic symbols like letters on a page of text or bricks on a wall and suggest that the duplex nature of the speech waveform is the core "defining feature" of robust human speech perception in noise and under adverse degraded listening conditions. In presenting evidence for this unconventional view of speech recognition, I also suggest that any principled theoretically-motivated account of human speech perception and spoken language processing will also have to be compatible with what we currently know about human information processing and how human memory and learning works. Speech perception and spoken language processing do not take place in isolation and they are inseparable from core memory and learning processes that reflect the operation of the whole system working together as an integrated process. To quote W. Nata, "no part of the brain functions on its own but only through the other parts of the brain with which it is connected." Put in a different way, it takes a whole brain to perceive speech-- the ear is connected to the brain and the brain is connected to the ear.

S8: EFFORTFUL LISTENING: PERCEPTION AND LEARNING OF DEGRADED SPEECH DEPENDS CRITICALLY ON ATTENTION

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In everyday life, people often hear speech that has been degraded (e.g., by background noise) or when they are distracted by other tasks. How well is that speech processed, and does processing under distraction depend on the quality of the signal? In a recent study [Wild, Yusuf, Wilson, Peele, Davis and Johnsrude 2012, J. Neurosci., 32, 14010-21] fMRI was used to address these questions, in young normally hearing listeners. On every trial, participants (n=21) attended to one of three simultaneously presented stimuli: a sentence, an auditory distracter, or a visual distracter. Sentences were presented at one of four acoustic clarity levels, including clear speech, 6-band noise-vocoded (NV) speech [Shannon et al., 1995, Science 270, 303-304], 6-band NV speech with compressed envelopes, and spectrally rotated 6-band NV speech (as an unintelligible control stimulus). A post-scan recognition test showed that clear speech was processed even when not attended, but that attention greatly enhanced the processing of degraded speech. Furthermore, speech-sensitive cortex could be divided according to how speech-evoked responses were modulated by attention, and these divisions mapped onto the hierarchical organization of the auditory system. Finally, frontal regions,including left inferior frontal gyrus, were only engaged when listeners were attending to speech and these regions exhibited elevated responses to degraded, compared with clear, speech. We suggest this response is a neural marker of effortful listening.

In another experiment, we examined whether learning to understand NV speech requires attention [Huyck and Johnsrude 2012, J. Acoust. Soc. Am.,131, EL236-42]. Young normally hearing participants (n=72) were randomly assigned to one of four groups. They either did not receive training (naïve), or during 15 consecutive trials, attended to one of three simultaneously presented stimuli: sentences, auditory distracters, or visual distracters (3 "attention" groups). When subsequently asked to report all the words they could understand from a new set of 20 6-band NV sentences (presented alone), intelligibility scores for the attention-to-speech group were significantly higher than those of a naïve (untrained) group. In contrast, report scores for the two distracter groups were indistinguishable from those of the naïve group. Attention to noise-vocoded speech appears to be required for learning.

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S9: HOW COGNITIVE FACTORS CAN AFFECT THE OUTCOME IN COCHLEAR IMPLANTS

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The outcome of cochlear implantation varies highly from one individual to the next. Some predictive factors, such as duration of deafness, have been readily identified. Recent research suggests cognitive ability to be another important factor, as speech comprehension requires both bottom-up acoustic analysis of speech signals and top-down linguistic and cognitive processes to extract meaning. Further, when the quality of the incoming speech signal is compromised due to cochlear implant (CI) signal transmission and/or poor listening conditions, the cognitive system needs to do extra work to compensate for the degradation. Thus, speech intelligibility with a CI is not only determined by the quality of the auditory input transmitted to the auditory nerve, but also by how well this input is utilized by the cognitive system of the individual CI user.

This talk will focus on two ways in which CI performance can be affected by cognitive factors, namely, perceptual restoration of degraded speech and changes in listening effort. In phonemic restoration, comprehension of degraded speech is enhanced by using context, linguistic rules and knowledge, and expectations. Our recent research has shown that hearing impairment and age, factors that affect both the quality of bottom-up signals and the cognitive skills, may counteract the benefit from restoration. These results are important in the case of CI users where the signal quality is affected by device signal processing and transmission, and cognitive skills by advanced age. Hence, it is not clear whether CI users can actively use the top-down processes to benefit from phonemic restoration, and in case the answer is no, whether they can be trained to do so. Our preliminary data imply that CI users may indeed have reduced or no benefit from restoration, but a targeted training may nevertheless increase their comprehension of degraded speech.

Comprehension of degraded speech also requires increased listening effort. This, in turn, could leave fewer cognitive resources available for concurrent tasks, such as remembering what was said, especially if the CI is not optimally fit. If sustained over long periods of time, increased effort can also lead to mental fatigue. Our recent research with CI simulations has confirmed that different CI settings may require differing levels of listening effort, even if they seemingly produce the same levels of speech intelligibility. We are now investigating how listening effort may play a role in performance with actual CI users, as well as various CI configurations, such as bimodal hearing. We expect that listening effort will be an even more important factor with actual CI users, who tend to be elderly, and that different CI configurations may provide additional benefits in terms of reduced listening effort.

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S10: PITCH, FUSION, AND BINAURAL SPECTRAL INTEGRATION

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There is significant variability in the benefit that hearing-impaired individuals receive from combining two hearing devices bilaterally, whether the devices are bilateral hearing aids (HAs), bilateral cochlear implants (CIs), or a CI worn with a HA in the contralateral ear (bimodal CI+HA). Some individuals experience useful improvements, but others receive little additional benefit or even interference with two hearing devices. One contributing factor may be an experience-induced change in the central processing of binaural inputs.

In normal-hearing (NH) listeners, the two ears provide essentially matched spectral information and thus sensory reinforcement to each other to average out noise. This process of integrating spectral information between ears can be called binaural spectral integration (BSI). In contrast, hearing-impaired listeners often have interaural pitch discrepancies due to hearing loss, i.e. diplacusis, or due to hearing device programming, e.g. CI programming, where real-world frequencies allocated to the electrodes differ from the electrically stimulated cochlear frequencies. Previous studies have shown that pitch perception adapts over time to reduce these discrepancies in some, but not all CI users (Reiss et al., 2007; 2011). Here we present new findings suggesting that BSI may also adapt to reduce the perception of interaural pitch discrepancy.

Our data show that individuals with long-term CI and/or HA experience have wider binaural fusion ranges than NH listeners, and can fuse inharmonic, dichotic tones that differ by as much as 2-3 octaves into a single percept. Consistent with studies of multi-input integration in other sensory modalities, these fused tones elicit a new binaural pitch that is an average of the monaural pitches, suggesting integration of mismatched rather than matched spectral information. This increased fusion and associated spectral averaging between ears indicates abnormal BSI in the hearing-impaired population, and may account for speech perception interference effects observed with binaural compared to monaural hearing device use.

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S11: WHY IS BINAURAL PERFORMANCE WITH BILATERAL COCHLEAR IMPLANTS POORER THAN EXPECTED? INSIGHTS FROM NEURAL DATA

Bertrand Delgutte, Kenneth Hancock and Yoojin Chung
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Although bilateral cochlear implants (CI) provide benefits in sound localization and for attending to the ear with the best signal-to-noise ratio, binaural performance of CI users is still far from normal, particularly in tasks involving interaural time differences (ITD) such as binaural unmasking. This poor performance is physiologically surprising because electric stimulation of the auditory nerve produces the strong phase locking that is essential for the operation of the brainstem circuits that create neural ITD sensitivity. Indeed, in animals with normal auditory development that are deafened and bilaterally implanted as adults, the coding of ITD in the auditory midbrain can be as precise as in normal hearing. Single-unit studies of the inferior colliculus (IC) suggest both deprivation of auditory experience and processor limitations contribute to this discrepancy.

In contrast to the animals used in previous studies of neural ITD coding, CI users often experience long periods of auditory deprivation before implantation and such deprivation can occur during a critical period for the experience-dependent maturation of binaural circuits. To determine the effects of deprivation, we compared ITD sensitivity of IC neurons between congenitally deaf cats that were implanted as adults and cats with normal auditory development that experienced either very brief (one week) or longer (6 months) periods of deafness. In congenitally deaf cats, the incidence of ITD sensitive neurons was greatly reduced compared to both groups of adult-deafened cats. Long-term, adult deafened cats showed a milder deficit in that the distribution of best ITD across the neural population lacked the contralateral bias that is present in both normal-hearing and acutely deafened cats, and is essential for fine ITD acuity near the midline. The finding that congenital deafness more severely degrades neural ITD coding than prolonged deafness in adulthood argues for the importance of fitting deaf children with sound processors that provide reliable ITD cues at an early age.

Even in cases when minimal prior deafness preserves good ITD sensitivity to low-rate electric pulse trains, neural and behavioral ITD sensitivities severely degrade for pulse rates above a few hundred Hz, where IC neurons only give an onset response. This is a problem for today's CI processors, which deliver ITD cues in the modulations of high-rate pulse trains. Using pulse train stimuli with irregular timing, we find that both sustained responses and ITD sensitivity can be restored in IC neurons by introducing short interpulse intervals. This finding suggests a new strategy for improving binaural performance in CI users by introducing additional pulses at select times to enhance perceptually important events.

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S12: EMERGENCE OF BINAURAL HEARING IN BILATERALLY IMPLANTED CHILDREN

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The standard of care in audiology is to provide bilateral cochlear implants (CIs) to children, and to many adults who have severe-to-profound bilateral deafness. Although bilateral input promotes the development of hearing in each ear, it is not clear whether integration of inputs from the two ears by neural mechanism is achieved in an ideal manner. It is noteworthy that many children with bilateral CIs display language and speech understanding performance within the normal range for their chronological age, however their ability to perform on spatial hearing tasks is typically under-developed. One source of limitation is in the signal processing and speech coding strategies utilized by today's clinical CIs, which do not coordinate the inputs from the two devices. Another issue to consider is that lack of any input may result in general neural degeneration, and lack of binaural input may result in degeneration of binaural circuitry.

This talk will focus on several issues. First, results will be presented from psychophysical studies aimed at understanding how binaural cues can best be restored to bilateral CI users. The issues to consider include the matching of inputs across the ears and role of spread of current with multi-electrode stimulation. Second, we consider the role of auditory plasticity due to congenital vs. later-onset of deafness. Some of our results suggest that bilateral CI users who experience acoustic hearing early in life are best able to utilize binaural cues, and that degradation of binaural processing is more profound for processing of timing cues than for level cues. Third, we consider the role of training and in the emergence of binaural processing and spatial hearing skills. In children with normal hearing we show that audio-visual integration may play a significant role in the ability of children to localize sounds in degraded listening conditions. In children who use CIs, the importance of training is especially relevant because spatial hearing is much less intuitive, and performance depends heavily on how well children can extract information from cues that are not presented in everyday listening through clinical processors. Finally, regarding top-down processing, measures of executive function (e.g., working memory and attention) have the potential to predict conditions under which speech understanding in complex listening situations will be most challenging.

Support provided by NIH-NIDCD (R01DC003083, R01DC008365).

S13: BILATERAL COCHLEAR IMPLANTATION CAN PROTECT THE DEVELOPING AUDITORY SYSTEM FROM UNILATERALLY DRIVEN REORGANIZATION

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Unilateral deafness restricts excitatory input to the auditory pathways from the deprived ear and removes inhibitory input involved in binaural processing. In developing animal models, this results in overexpression of neural connections from the hearing ear and compromises binaural responses. In the present study, we asked whether such reorganization occurs when children with bilateral deafness hear through unilateral cochlear implants and, if so, whether providing bilateral electrical input in early development protects them from such changes.

Cortical activity was measured by electroencephalography in: a) 34 children with early onset (typically congenital) deafness and implanted with unilateral or bilateral cochlear implants at <3 years of age and b) 7 peers with normal hearing. Twenty-four children received a right unilateral cochlear implant. Sixteen of this group later received a second cochlear implant in the opposite ear (inter-implant delay=3.57±0.72 years). Another group of 10 children received bilateral implants simultaneously. Bilateral cochlear implant use was at least 30 months at time of testing. Our TRACS beamformer was used to suppress cochlear implant artifact and locate evoked activity in auditory cortices.

Right sided cochlear implant use increased dipole moments in the left and right auditory cortices whereas simultaneous bilateral cochlear implantation promoted normal levels of activity. Abnormal increases in activity from the stimulated right ear were significant by > 1.5 years of cochlear implant use and were not reversed even after > 30 months of bilateral implant use.

The data suggests that bilateral input must be provided within a sensitive period to the developing human auditory system.

S14: SPATIAL ATTENTION IN BILATERAL COCHLEAR-IMPLANT USERS

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In complex auditory scenes such as a cocktail party, people with normal binaural hearing have the ability to focus auditory attention on a single sound source and ignore the competing sound sources, which produces exceptional speech understanding advantages compared to those without normal binaural hearing. These advantages derive from listeners being able to selectively attend to the ear with the better signal-to-noise ratio produced by “head-shadow” and being able to access binaural difference cues that facilitate stream segregation for spatially-separated sounds. However, people with bilateral cochlear implants (BICIs) often struggle to understand speech in noisy environments, and derive few binaural benefits beyond better-ear listening. Here we utilized a testing paradigm developed by Cherry [J. Acoust. Soc. Am., 25, 975-979 (1953)] that eliminates better-ear listening in an attempt to understand spatial attention and binaural stream segregation in BICI listeners. We hypothesized that (1) BICI listeners can derive better-ear benefits comparable to normal-hearing (NH) listeners and that (2) BICI listeners can derive spatial hearing benefits by presenting the target talker to one ear and maskers to both ears, a spatial configuration where the target should be perceived as spatially separated from the perceptually centered masker by an unrealistically large (i.e., infinite) interaural level difference.

We measured speech reception thresholds (SRTs) for 11 BICI listeners by presenting sentences through the direct audio input of the CI processors. Listeners reported the words of a female target (T) while attempting to ignore a male masker (M). Five conditions were tested: T in one ear only (in quiet), T in attended ear and contralateral M, monotic T and M, diotic T and M, and monotic T and diotic M. NH listeners were tested using unprocessed and vocoded speech for comparison data.

Comparing the contralateral masker to the quiet conditions, SRTs worsened by ~8 dB for both BICI and NH listeners, supporting the first hypothesis that BICI listeners have comparable better-ear listening to NH listeners. Comparing the monaural T and diotic M to the monotic/diotic T and M, SRTs improved by 7 dB for NH listeners, however, SRTs did not change in for BICI listeners. Therefore, contrary to the second hypothesis, the data suggest that BICI listeners cannot derive spatial hearing and stream-segregation benefits for even the most extreme difference in spatial configurations. These data suggest possible central limitations in obtaining further binaural benefits for understanding speech in complex auditory environments.

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S15: DUPLEX PERCEPTION BY LISTENERS WITH BILATERAL COCHLEAR IMPLANTS: A MEASURE OF BILATERAL INTEGRATION OF SPEECH

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Bilateral cochlear implantation does not confer all the advantages of the normal binaural auditory system because outputs from the processors are not synchronized in time or frequency. Successful use of bilateral cochlear implants (BiCIs) is likely limited by a listener's ability to combine bilateral inputs into a coherent percept. In this project we propose a method to test for integration of speech information across ears for BiCI listeners.

Previous tests of integration of dichotic speech sounds have suffered from the limitation that plagues most traditional speech tasks; redundancy in the signal permits a listener to correctly identify the target sound using either ear in isolation, even if it contains "holes" or other degradations in the spectral domain. Thus, correct identification of most dichotic speech signals is possible without any integration at all.

In this study we use a perceptual task that can be used to specifically test for bilateral integration in speech sounds, without the risk of either ear contributing solely to the perception. Duplex perception is a process by which a monotic speech syllable is digitally modified to become ambiguous via omission of a selected spectral region that is crucial to a phonetic contrast. That crucial region is delivered to the contralateral ear and is by itself not a recognizable speech sound; the complete sound is identified correctly only if auditory information is integrated across the ears.

In preliminary results using 10 listeners with BiCIs and 10 normal hearing listeners in ACE-style vocoder simulations, we tested performance in Duplex perception tasks where incoming speech information was integrated across various targeted frequency regions. In this presentation we show patterns of variability in BiCI listeners, and discuss the relationship between duplex perception and other binaural hearing skills. The results of this experiment will help to explain the basic mechanism of bilateral integration for BiCI listeners as it applies to speech sounds, and also provide a diagnostic tool to identify listeners who are struggling with BiCIs and/or those who may benefit from unconventional dichotic coding strategies.

This work was supported by NIH-NIDCD Grant No. R01 03083 (PI: Ruth Y. Litovsky)

S16: LONG-TERM EFFECTS OF NEUROTROPHINS AND ELECTRICAL STIMULATION ON SURVIVAL AND FUNCTION OF COCHLEAR SPIRAL GANGLION NEURONS

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The postnatal development and survival of cochlear spiral ganglion (SG) neurons depend upon both neurotrophic support and neural activity. Our studies in cats deafened prior to hearing onset have shown that intracochlear electrical stimulation (ES) from a cochlear implant (CI) can partly prevent SG degeneration, and intracochlear infusion of brain-derived neurotrophic factor (BDNF) can further improve neural survival. Kittens were deafened as neonates (systemic neomycin) and implanted at ~1 month of age with custom-designed scala tympani electrodes with an integral drug-delivery cannula connected to a mini-osmotic pump. BDNF was infused for 10 weeks, and ES continued for 5-6 months. Combined BNDF+ES elicited markedly improved SG survival (>50% increase) and larger cell areas in treated ears compared to contralateral. Further, SG survival after BDNF+ES was significantly higher than after ES alone over these prolonged periods. Other interesting findings after BDNF+ES included angiogenesis in the SG, increased peri-implant fibrosis, higher density/larger size of radial nerve fibers in the osseous spiral lamina, ectopic sprouting of fibers into the scala tympani, and longitudinal improvement in EABR thresholds (for chronically stimulated CI channels only). In a further study, we used identical deafening and BDNF infusion procedures, but no subsequent ES was delivered. In animals studied at ages similar to the BDNF+ES group, SG data showed that neurotrophic effects persisted for >3 months after terminating BDNF. Numerous sprouted radial nerve fibers again were observed in the scala tympani. Terminal electrophysiological recordings in the inferior colliculus (IC) in these same animals showed that the basic cochleotopic organization was preserved despite early deafness. Lower IC thresholds were correlated with more selective central activation after ES alone, and in deafened controls, but not in BDNF-treated animals, in which variability was much greater, perhaps due to the ectopic radial nerve fibers.

Improved survival of SG neurons and radial nerve fibers, as well as lower EABR thresholds, may be beneficial for CI outcomes; but disorganized, ectopic radial nerve fiber sprouting could be deleterious to optimized CI function. Further, neurotrophin (NT) delivery by osmotic pumps is not practical for clinical application. Therefore, our recent studies have explored the possibility of using adeno-associated viral vectors (AAV) to force NT expression by cells in the cochlea. Preliminary immunohistochemistry data obtained 2 weeks after AAV2-GFP (green fluorescent protein) injections into newborn mouse cochlea and 6 weeks after injections into neonatally deafened cat cochlea showed robust expression of the GFP reporter gene in hair cells (where present), inner pillar cells and in SG cells throughout the cochlea. Further, injections of an AAV2-NT (human GDNF) into mouse cochlea elicited robust expression of human GDNF, as demonstrated by preliminary qPCR data. We hypothesize that AAV2-NT transfection will elicit NT expression at more physiological levels than osmotic pumps and thereby promote improved SG survival and more normal radial nerve fiber neuroanatomy.

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S17: LOSS OF TEMPORAL FIDELITY IN THE AUDITORY PATHWAY FOLLOWING AUDITORY DEPRIVATION

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Ultimately, success in restoration of hearing with an auditory prosthesis is limited by the condition of the auditory nerve and its central projections. In the present study, we deafened adult cats and waited 6-9 months to permit some loss of auditory nerve fibers and a period of reduced auditory neural activity. Then, in terminal experiments, we assessed transmission of spectral and temporal information from two types of auditory prosthesis, through the auditory brainstem, to the central nucleus of the inferior colliculus (ICC). The auditory prostheses were a banded intrascalar cochlear implant and a penetrating intraneural electrode array. Our previous work in acutely deafened animals has shown that, compared to conventional intrascalar stimulation, intraneural stimulation provides more restricted activation of frequency-specific auditory nerve populations at dramatically lower excitation thresholds (Middlebrooks and Snyder, JARO 8:258-279, 2007). Also, intraneural stimulation permits transmission of temporal fine structure at higher frequencies (Middlebrooks and Snyder, J Neurosci 30:1937-1946, 2010). In the present terminal experiments, we recorded at 32 sites along the tonotopic axis of the ICC while stimulating the ear electrically. Control animals had normal hearing until they were deafened on the day of the terminal experiment.

Post-mortem histological examination demonstrated ~30% loss of auditory nerve fibers in our chronic hearing impaired animals. Measures of transmission of spectral (i.e., cochlear place) information showed little effect of chronic hearing impairment. Thresholds for excitation with single electrical pulses and tonotopic spread of activation were within control ranges for both types of stimulation. In contrast, temporal fidelity was markedly degraded in chronically deaf animals. In the control (acutely deaf) animals, 50% of ICC neurons showed significant phase locking to intraneural electrical pulse trains at rates of ≥205 pulses per second (pps), and 37% of ICC neurons responded at that rate to cochlear-implant stimulation. In the chronically deaf cats, fewer than 15% of ICC neurons phase locked at 205 pps.

The observed loss of temporal fidelity might be due in part to a reduction in absolute numbers of nerve fibers. Nevertheless, we hypothesize that a greater effect is the chronic loss of synchronous synaptic activity in central neurons during the 6+ months of reduced auditory nerve activity. We are testing the hypothesis that restoration of synchronous activity by electrical stimulation might reverse the observed degradation of temporal fidelity.

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S18: DRUG DELIVERY TO THE COCHLEA FOR NEURAL SURVIVAL

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Auditory neurons, the target cells of the cochlear implant, undergo progressive degeneration following deafness, ultimately leading to cell death. The delivery of drugs to the cochlea, such as neurotrophins, has been shown to protect auditory neurons and promote regrowth of their peripheral fibres. However, an effective strategy to safely deliver therapeutics is yet to be established thus limiting the clinical potential of this approach.

This talk will describe experiments that have examined the use of two different clinically applicable approaches (gene therapy and nanotechnology) that aim to deliver drugs to the cochlea for long-term auditory neuron protection. In these experiments we have used adult guinea pigs that were deafened with aminoglycoside drugs to cause a profound sensorineural hearing loss. In one set of experiments animals received treatment with gene therapy with an inoculation of an adenoviral vector containing genes for neurotrophins or an inoculation with a control vector. In another set of experiments, animals were implanted with nanoengineered particles. One ear received particles loaded with neurotrophins and the other ear received control particles. Both approaches resulted in significant auditory nerve survival.

The preservation and regrowth of auditory neurons may lead to improvements in clinical outcomes for cochlear implant recipients when combined with new electrode arrays and processing strategies that take advantage of greater neural populations and improved electrode-neural interface. In addition to neural protection and regrowth, the safe and effective delivery of therapeutic drugs to the inner ear may also enable the preservation or restoration of residual sensory function that is known to deteriorate following cochlear implantation. The application of drugs to protect and maybe even restore both neural and sensory elements is likely to be a key factor in improved clinical outcomes for cochlear implant recipients in the future.

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S19: CHANGES IN CORTICAL RESOURCE ALLOCATION IN HEARING LOSS

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A basic tenet of neuroplasticity is that the brain will re-organize following sensory deprivation. Sensory deprivation appears to tax the brain by changing its normal resource allocation. A better understanding of the cortical re-organization that accompanies hearing loss may allow us to incorporate improvements in the design of prostheses to allow them to better accommodate altered cortical processing. Compensation for the deleterious effects of hearing loss may include recruitment of alternative or additional brain networks to perform auditory tasks. Our high-density EEG experiments suggest that adult-onset hearing loss results in significant changes in neural resource allocation, reflecting patterns of increased listening effort and decreased cognitive reserve, which may be associated with dementia-related cognitive decline. Cross-modal plasticity is another form of cortical re-organization associated with deafness. Cross-modal plasticity occurs when an intact sensory modality recruits cortical resources from a deprived sensory modality to increase its processing capabilities as compensation for the effects of sensory deprivation. Deaf animals show recruitment of higher-order auditory cortical areas, by visual and somatosensory modalities resulting in enhanced capabilities for the recruiting modality, with likely negative consequences for auditory perception. Our results in humans suggest clear evidence of recruitment of higher-order auditory cortical areas by visual and somatosensory modalities in both congenital deafness and adult-onset hearing loss. Cross-modal cortical re-organization is evident in early stages of hearing loss and shows a strong negative correlation with speech perception outcomes, suggesting that recruitment by visual and/or somatosensory modalities may influence the variability in outcomes seen in implanted patients. Overall, our results suggest that compensatory cortical plasticity secondary to sensory deprivation has important neurological consequences and influences outcomes for hearing-impaired individuals.

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S20: CROSS-MODAL PLASTICITY IN THE CONGENITALLY DEAF CAT

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Congenital and early deafness reorganizes the auditory system and leads to supranormal abilities in other sensory systems. Two different mechanisms for such reorganization have been suggested: (i) recruitment of the deprived auditory structure by another sensory system, i.e. vision; (ii) reorganization of attentional resources towards the other sensory systems. In a previous study we could demonstrate that the field A1 (primary auditory field) is an unlikely site of cross-modal reorganization (Kral et al., 2003, Exp Brain Res). However, in behavioral experiments we showed that congenitally deaf cats demonstrate supranormal performance in the visual domain (detection and localization in the periphery and movement detection, Lomber et al., 2010, Nat Neurosci). Furthermore, cooling deactivation localized the site of reorganization into secondary fields dorsal zone (DZ) and posterior auditory field (ibid.).

To investigate the neuronal mechanism of such reorganization, the follow-up experiments focused on field DZ. Application of retrograde tracer into DZ and A1 demonstrated a differential recruitment of these fields in cross-modal reorganization in congenitally deaf cats: whereas A1 did not show significant cortical cross-modal reorganization, field DZ showed visual and somatosensory projections not observed in hearing controls (Barone et al., 2013, PLoS One). The reorganization was, however, only modest: only few % of all projections were identified as new crossmodal projections.

In the present experiments undertaken on 8 cats (4 deaf, 4 hearing), recordings were taken from field DZ and an adjoining visual field (PMLS) under cochlear implant (auditory) and visual stimulation in precisely-controlled isoflurane/N2O anesthesia. Visual stimuli were flashes (increasing and decreasing luminance) and, to include motion, phase-reversal gratings with different orientations and spatial frequencies. Simultaneous recordings were taken using two 16-channel Neuronexus probes inserted at ~10 positions in both fields spatially as close as possible. The DZ penetration had an orientation parallel to the microcolumns. Electrodes were stained with DiI or DiO and the penetrations were histologically reconstructed after the experiment. Recording sites within DZ were confirmed using an SMI-32 staining.

From all recording sites, only the sites containing unit activity were processed further. Alltogether, 1424 unit responses were evaluated, ~700 in each group. Spontaneous activity was significantly higher in PMLS than in DZ in both groups of animals. Unit responses showing statistically significant correlation with current level were considered responsive to auditory stimulation. There were significant auditory responses in both groups of animals in DZ (deaf: 29%; hearing: 36% of sites). In PMLS, only few units responded to auditory stimulation (deaf: 2%; hearing: 4%). Visual stimulation led to responses in one third of units in PMLS in both groups of animals (deaf: 31%, hearing: 29%). In DZ, on the other hand, the deaf cats had more visual responses (deaf: 7%, hearing: 2%). In these responsive units, the evoked firing rate was higher in deaf cats, whereas hearing cats demonstrated more a modulation of ongoing activity than an evoked response in DZ. Only few bimodal units responsive to both auditory and visual stimulation were observed in DZ of deaf cats.

The present results suggest a modest visual cross-modal reorganization of field DZ in congenital deafness. However, the auditory responsiveness was preserved in this field, too. The relatively small number of visually-responsive cells, corresponding to previous tracer studies (Barone et al., 2013), demonstrates that also other (possibly attentional) processes are involved in supranormal visual performance of deaf subjects.

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S21: MULTIMODAL SENTENCE RECOGNITION BY CHILDREN WITH COCHLEAR IMPLANTS OR HEARING AIDS

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Under natural conditions, listeners use both auditory and visual speech cues to extract meaning from speech signals containing many sources of talker variability. However, traditional clinical tests of spoken word recognition routinely employ isolated words or sentences produced by a single talker in an auditory-only presentation format. The more central cognitive processes used during multimodal integration, perceptual normalization and lexical discrimination that may contribute to individual variation in spoken word recognition performance are not assessed in conventional tests of this kind. We have developed new tests of multimodal sentence recognition that are theoretically motivated by the Neighborhood Activation Model of spoken word recognition. The Multimodal Lexical Sentence Test for Children (MLST-C) contains sentences with key words that are independently controlled for the lexical characteristics of word frequency (i.e., how often a word occurs in the language) and lexical density (i.e., the number of phonemically similar words, or neighbors, to the target word). To incorporate the kind of variability that listeners encounter in the real world, the sentences were drawn from stimuli produced by five male and five female talkers. The tests can be administered in a visual-only (V), auditory-only (A) or auditory-plus-visual (AV) presentation format.

To date, we have tested approximately 70 children with this new sentence recognition measure. Participants include both pediatric cochlear implant recipients and children with mild-to-moderately-severe hearing loss who use bilateral hearing aids. All children were administered several different MLST-C sentence lists in each of the three presentation formats. Auditory stimuli were presented at 60 dB SPL in quiet. A repeated measures analysis of variance revealed a significant group by presentation format interaction. Within the cochlear implant group, sentence recogntion differed significantly in all three formats; scores were lowest in the V presentation format followed by the A and AV formats, respectively. The hearing aid users' sentence recognition performance was significantly poorer in the V presentation format than in either the A or AV formats, but their scores did not differsignicantly between the latter two presentation formats. Between-group comparisons revealed that the cochlear implant and hearing aid groups did not differ in visual-only sentence recognition. However, sentence recognition was significantly better for the hearing aid group in both the A and AV presentation formats.

We are currently conducting additional analyses to examine talker and lexical effects on spoken word recognition by our participant groups in each presentation format. The results of these analyses will be discussed in this presentation.

S22: FUNCTIONAL NEAR INFRARED SPECTROSCOPY REVEALS CORTICAL CROSS-MODAL REORGANISATION FOLLOWING DEAFNESS

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Studies using neuroimaging, including functional positron emission tomography [Rouger et al., 2011, Hum Brain Mapp. 33:1929-1940], have suggested a link between deafness-induced cortical cross-modal activation patterns and cochlear implant performance. Near infrared spectroscopy (NIRS) is a relatively novel functional-neuroimaging technique that has been used successfully to assess auditory-evoked cortical responses in deaf children with cochlear implants [Sevy et al., 2010, Hear Res. 270:39-47] and is unaffected by electrical artefacts or the implant magnet. We studied cortical responses to multi-sensory stimuli (auditory, visual and somatosensory) using NIRS in 28 normal-hearing (mean age = 34 years ±13 years stdev, n=16 female) and 16 profoundly-deaf adult participants (mean age = 43 years ±10 years stdev, n=7 female).

Auditory stimuli consisted of sinusoidal amplitude-modulated (10Hz; 100% modulated) or unmodulated broadband-noise bursts presented at 80dB SPL. Visual stimuli consisted of 1000 white dots on a black background that were randomly flashing or coherently moving. Somatosensory stimuli were 10Hz or 20Hz sinusoidal vibrations, delivered to the palms of both hands. All stimuli had duration 20s and were presented five times each in a pseudorandom order, interleaved with rest periods of random duration (25-45s). NIRS data were acquired from the temporal lobes using a 24-channel Hitachi ETG4000 system. Data were processed using the NIRS software suite, HomER2.

Oxygenated and deoxygenated haemoglobin changes were calculated from raw optical intensity data, and averaged across trials of the same stimulus. A response was defined as a significant increase or decrease in concentration of either chromophore that were correlated temporally with stimulus presentation. The majority of normal-hearing participants (82%) exhibited responses to at least one auditory stimulus, while responses to visual and somatosensory stimuli were observed in just 4% and 7% of participants, respectively. Only a small proportion (13%) of the profoundly-deaf group responded to sound, while about half responded to visual (56%) and somatosensory (50%) stimulation. Future studies will use this paradigm to measure multi-modal activation of auditory cortex in profoundly-deaf individuals before and after cochlear implantation. We hypothesise that cross-modal cortical plasticity assessed using NIRS may provide a useful prognostic indicator of cochlear implant outcome.

This work is supported by the University of Nottingham, the MRC and the NIHR.

S23: PHILIP LOIZOU – AN APPRECIATION

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Philip Loizou -- my student, my colleague, my teacher, my friend -- died last summer at the age of 46. In this talk I will review some of his personal history and professional accomplishments. Philip received his B.S., M.S., and Ph.D. degrees from Arizona State University in Electrical Engineering. He was my postdoctoral Fellow and was introduced to cochlear implants in 1995 - 1996. . Following a brief stay at the University of Arkansas at Little Rock, he joined the Department of Electrical Engineering at the University of Texas, Dallas where he helped co-found the Center for Robust Speech Systems (CRSS) in the Jonsson School and directed the Speech Processing and Cochlear Implant Labs within CRSS. He served as Professor of Electrical Engineering and held the Cecil and Ida Green Chair in Systems Biology Science at UT Dallas. Philip was an excellent scientist, teacher and mentor. He was 'old school' in the best sense of the term: His family was more important than his work; he was generous with his time to his colleagues and students and he never thought his skills to be extraordinary – he was a modest man. He leaves a wonderful legacy for those of us who were fortunate to know him well.

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S24: SPEECH PROCESSING OPTIMIZATION IN COCHLEAR IMPLANTS BASED ON OBJECTIVE MEASURES

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The optimization of speech processing strategies and the programming of cochlear implants (CI) relies on behavioral testing and active participation of the recipient. This might be greatly facilitated by using objective measures.

Auditory stimuli evoke increased neural activity at different stages of the auditory path. These give rise to onset, offset, transient and steady state responses depending on the specific stimuli used. This also applies for electrical stimulation in cochlear implants (CI). The responses can be registered in the EEG. However, the analysis of these neural responses in the EEG-recordings is limited by the occurrence of electrical artifacts linked to the CI-stimulation, particularly for steady state responses. In the last few years methods have been developed to reliably extract the steady state brain oscillations.

In this contribution we will report on auditory steady state brain responses (ASSR) evoked by electrical stimuli with different modulation frequencies, and their possible use to optimize the speech processing in cochlear implants. Depending on the modulation frequency from about 100 Hz to a few Hertz, the ASSR neural responses are generated at brainstem level to auditory cortex, respectively. Two hypotheses are investigated.

First, ASSRs to stimuli modulated at 4-20 Hz are related to the sensitivity to the speech envelope modulation frequencies and a model for optimal coding of the envelope, and thus speech understanding. Second, ASSRs to stimuli modulated at higher frequencies, with response generation at brainstem level, can be used to assess stimulation thresholds and dynamic range in sleeping and/or very young cochlear implant patients.

S25: SIGNAL OPTIMIZATION FOR COCHLEAR IMPLANTS: PHILIP LOIZOU'S CONTRIBUTION TO COCHLEAR IMPLANT RESEARCH

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Due to the loss of fine spectral details, cochlear implant (CI) patients have great difficulty understanding speech in challenging listening conditions. While increasing the spectral resolution and/or reducing channel interactions might improve performance in noise, optimizing the acoustic input in response to the acoustic environment or to different speaker characteristics before CI signal processing might also be beneficial, and make CI processing more efficient. This "signal optimization" approach would serve as an independent, front-end, pre-processing scheme and would thus be suitable for all CI users. Such approach has been one of the main research themes in both Philip's lab and my lab.

One "pre-processing" approach studied by my group and Phillip's group involves optimizing the input acoustic signal in response to the acoustic environment. Existing speech enhancement algorithms can improve speech quality but not speech intelligibility. Philip and colleagues recently developed a theoretical framework that can be used to analyze factors that may influence the intelligibility of processed speech. Analytic data suggest that certain distortions, such as amplification distortion, are more detrimental to speech intelligibility than others. Large gains in intelligibility were obtained by human listeners when these distortions were properly controlled. Work by Philip's lab and my lab has demonstrated that appropriate speech enhancement algorithms may improve CI users' speech understanding in background noise.

A second pre-processing approach involves optimizing the input acoustic signal in response to the different speaker characteristics. Some speech signals can be maybe difficult to understand even under quiet listening conditions if they strongly deviate from familiar patterns. Normal-hearing (NH) listeners can readily understand speech, despite large variations in talker characteristics and language context. This phenomenon of "phonetic constancy" suggests that some perceptual adjustment process helps to normalize acoustic variability across patterns. This "speaker normalization" process may depend on fine spectro-temporal details that CI users cannot access due to the limited spectral resolution with their device. As such, CI users are more susceptible to differences in acoustic patterns between talkers. Several attempts have been conducted to optimize the "accent" input acoustic signals using signal processing approaches.

One of the biggest projects from Philip's group was to develop a PDA-based research platform for implementing novel speech processing strategies and conducting psychophysical experiments. Not surprisingly, the project encountered difficulty implementing the front-end to the different CI devices. Nevertheless, the concept of a flexible front end that would allow for long-term studies with experimental processors remains a compelling challenge. Hopefully, this project will continue in some form to expand research opportunities and ultimately, to improve optimization of CI signal processing for individual patients and/or listening conditions.

S26: SOUND PROCESSING TO IMPROVE TRANSMISSION OF INTERAURAL LEVEL DIFFERENCES WITH COMBINED COCHLEAR IMPLANT AND HEARING AID STIMULATION

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Due to changing implantation criteria, many recently implanted cochlear implant users have residual hearing in the non-implanted ear, which is stimulated using a hearing aid. This combination is called bimodal stimulation. Currently devices are used that were developed separately and are often fitted separately. This results in very different growth of loudness with level in the two ears, potentially leading to decreased wearing comfort and suboptimal perception of interaural loudness differences. To address this, we developed a loudness equalization strategy and fitting method named 'SCORE bimodal'. The strategy aims to equalize loudness growth for both modalities using existing models of loudness for both acoustic and electric stimulation, and is suitable for implementation in wearable devices.

We performed loudness balancing experiments with six bimodal listeners to validate the strategy. In a first set of experiments, monaural loudness balancing was done for four harmonic complexes of different bandwidths, ranging from 200 Hz to 1000 Hz. Both the electric and acoustic loudness model predicted the psychophysical data well. In a second set of experiments, binaural balancing was done for the same stimuli. It was found that on average SCORE significantly improved binaural balance by 59 percent.

Speech perception performance in quiet and in noise, and sound localization ability of the same six bimodal listeners were measured with and without application of SCORE. Speech perception in quiet was measured with only acoustic, only electric, or bimodal stimulation, at soft and normal conversational levels. There was a significant improvement in phoneme score with application of SCORE. Speech perception in noise was measured with a steady-state noise, a fluctuating noise, and a competing talker, at conversational levels for bimodal stimulation. There was no significant effect of application of SCORE.

As SCORE was found not to interfere with speech perception and to improve loudness perception, it seems suitable for implementation in clinical devices. We believe that this is the first sound processing strategy and fitting method developed specifically for bimodal stimulation.

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S27:USING THE ESTIMATED ELECTRODE-NEURON INTERFACE TO OPTIMIZE SPEECH PROCESSOR SETTINGS IMPROVES SPEECH PERCEPTION

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Cochlear implant programming in clinical devices has evolved from focused stimulation using the bipolar configuration to that of the monopolar mode, mostly to reduce power consumption. As technology has advanced, we have come full circle back to exploring the possibilities of stimulating an implant with focused electrical fields.

In the present study, the application of focused stimulation in sound processing strategies was explored. For each CI listener, channels with relatively high tripolar thresholds were presumed to have a poor electrode-neuron interface as a result of reduced localized neural viability or poor electrode positioning (i.e., electrodes on the lateral wall or traversing out of the scala tympani). Computerized tomography was used when available to assess the electrode position. Using the BEPS+ software, processing strategies were changed from the listener's everyday program to each of the following test conditions; 1) a 15-channel map with all channels programmed in the monopolar mode; 2) a 15-channel map with channel 1 programmed in monopolar and channels 2 through 15 programmed in the partial tripolar configuration (Ïf = 0.875 (highly focused)); 3) similar to map 2, but with two or three non-neighboring channels with high tripolar threshold deactivated; and 4) similar to map 2, but with two or three non-neighboring channels with low tripolar thresholds deactivated. Adult, post-lingually deafened users of the Advanced Bionics, HiRes90k cochlear implant participated in this study. All subjects had at least 6 months of experience with their cochlear implant. Listeners were given 10 minutes watching a television program with captioning using each strategy. Speech perception abilities were measured in the sound field with the AzBio sentences in quiet and multi-talker babble noise, and with medial vowel and medial consonant identification. Speech was presented at 60 dBA.

Results showed that most subjects performed best on sentences in noise when using the program with deactivated high-tripolar threshold channels compared to the other strategies. In the one subject for whom sentence testing in noise was too difficult, performance on vowels and consonants was best with the optimized speech processing strategy (focused with three channels with high threshold deactivated). These findings suggest that speech perception performance can be improved in long-term cochlear implant listeners by eliminating channels for which the underlying electrode-neuron interface is suspect.

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S28: SYNTHESIS-DRIVENNOISE SUPPRESSIONSTRATEGIES BASED ON HARMONIC REGENERATION

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Speechrecognitioninbackgroundnoiseisstillachallengingtask forcochlear implant(CI)listeners.Althoughasapre-processing based approach,noisereductionhas beenshowntobeeffectiveinimprovingCI users'speechperceptioninnoise,itoffersnoremedy forthe problemthat thenoise-suppressedspeechnolonger possessestheproperties man- dated bythe acousticprocess ofspeechproduction.

The main rationale todevelopsynthesis-drivennoise-reduction strategiesbased onharmonicregeneration isthatmostnoisereduction methodsusepresetmathematicaloptimizationcriteriathatdonottarget perceptual benefits,andas aresultsomebeneficial cuesare distortedor missing inthenoise-reducedspeech(e.g.,thebeneficialcuesoftheharmonic structuresforspeechrecognitionin maskers are usually severely distortedinthevoicedsegmentsofthenoise-suppressedspeech).Asa firststeptotacklethis problem,wepropose noise-reductionmethodsfor CIsusingthelog-minimummeansquareerror(LogMMSE)criteriacombined withstatistical-approach-based harmonic-regenerationalgorithms thatattempttoestimatethefundamentalfrequenciesandmagnitudesof theharmonics inthevoicedspeechsegments, andwewill evaluateseveral efficientalgorithms forsynthesizing harmonicsfromdegradedspeech, suchasthosebasedonadaptivecomb filtering andthoseemploying non- linear functions.Byregenerating theharmonicstructuresinthenoise-suppressedspeech,wehypothesizethat the beneficialcuesderived from theharmonicswillbeenhancedforimprovedspeechrecognitionperformanceinmaskers bythecochlear implantlisteners.

Inthepresentstudyinvolving bothtestingCIsubjects andtesting normal-hearing listeners using acousticsimulationimitatingelectroacoustic stimulation, we will examine and report subjects' performance in speechrecognitionin noiseormusicand discusshowsynthesis-driven noise-reduction strategies based on harmonic regeneration benefits speechperceptioninmaskersbycochlear implantusers.

S29: ACOUSTIC REVERBERATION IN COCHLEAR IMPLANT PROCESSORS WITH TWO MICROPHONES

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Cochlear implant (CI) users can identify words and sentences with fairly high accuracy in quiet and anechoic environments. However, their performance degrades significantly in reverberant environments. In fact, our prior research has shown that speech intelligibility by cochlear implant users in reverberation degrades exponentially as reverberation time increases. The simultaneous use of a cochlear implant (CI) and a hearing aid (HA) in the contralateral ear, or bimodal fitting, provides recipients with potentially complementary information to the auditory system for speech perception under reverberant listening conditions.

Such complementary information may be useful not only in quiet listening contexts but also in situations where the auditory information is corrupted by reverberation. In this talk, we will share recent data on speech and consonant perception under reverberation obtained with listeners who rely either on a cochlear implant alone or on bimodal fittings. We will also shed some light into the phonetic contribution of acoustic hearing by conducting a comprehensive analysis and assessment of the effects of reverberation on the linguistic cues of consonants, e.g., manner and place of articulation, as well as, voicing.

Finally, we will review prominent reverberation suppression strategies that could improve speech intelligibility in moderate-to-high reverberation times. We will focus mainly on strategies that rely on a dual-microphone setup. The implications of our preliminary findings on the rehabilitation of cochlear implant and bimodal users will also be discussed.

Research supported by NIDCD/NIH.

S30: REHABILITATION STRATEGIES BASED ON ACROSS-SITE PATTERNS OF TEMPORAL ACUITY

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Low frequency temporal envelopes of band-pass filtered speech signals carry one of the most important codes that enable speech recognition by cochlear implant (CI) users. The lowest modulation depth at which the subject can discriminate a modulated from an unmodulated pulse train, the modulation detection threshold (MDT), is predictive of speech recognition. Importantly, MDTs vary across stimulation sites in individual subjects and the across-site pattern (ASP) of MDTs is different for each subject, suggesting that the MDTs depend on conditions near each of the stimulation sites in the electrode array and that the conditions at any given location along the electrode array are subject specific. Most other psychophysical measures also show across-site variation but the ASPs vary from measure to measure suggesting that the mechanisms underlying the various psychophysical measures are not the same.

Across–site patterns of MDTs are usually stable over time and thus they can be used to identify specific areas along the electrode array that warrant treatment. We hypothesized that improving the across-site mean (ASM) MDT by selecting the sites with the best MDTs or improving MDTs at weaker sites, would result in improved speech recognition. This hypothesis was tested using a within-subject design that compared two 10-site experimental processor maps: one with the 10 best-MDT sites and one with the 10 worst-MDT sites from each of 5 frequency regions. The MDTs in this case were measured in the presence of an unmodulated masker on an adjacent site. The best-MDT map was usually better for recognition of consonants and vowels and it was better for recognition of sentences in noise in all subjects.

In a more clinically applicable procedure, an experimental map was created by turning off one site with the poorest MDT from each frequency region and reallocating the frequencies from the sites that were turned off. Compared to the subject's everyday-use map, performance was better with the experimental map for consonant and sentence recognition in noise but it was worse for vowel recognition, possibly due to the reduction in spectral resolution. In subjects with bilateral implants, site selection can be achieved without frequency reallocation by turning off poor sites in one ear and presenting the turned-off frequency information via the corresponding sites in the contralateral ear, thus preserving spectral resolution. Subjects showed improved recognition of vowels, consonants and sentences in noise with this approach.

Another approach for improving the ASM MDT is to adjust stimulation parameters on sites with poor MDTs. In a recent experiment we improved MDTs at the 5 poorest-MDT sites by artificially raising the T-level settings for those sites in the speech processor. This procedure improved MDTs, assessed at 20% of the dynamic range, and improved speech recognition relative to that when there was no level adjustment.

These site-specific strategies typically result in a few dB of improvement in the signal to noise ratio at which subjects can recognize speech signals in noise, despite minimal experience with the experimental processor settings. The improvements are small relative to across-subject differences but they are appreciated by the subjects.

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S31: TEMPORAL INFORMATION PROCESSING: CAN CURRENT FOCUSING HELP?

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Fine temporal structure appears to play an important role in auditory processing in normal hearing, but there is currently little evidence that cochlear implant users can utilise this information to improve speech understanding. A possible reason for this difficulty is the poor place specificity of electrical stimulation, leading to the inability to independently transmit fine temporal information at different cochlear locations. Another reason may be that the auditory system is attuned to specific time delays (or synchronies) at different cochlear places, again difficult to produce using electrical stimulation with poor place specificity. A series of experiments tested whether current focussing may improve perception of within- and across-electrode temporal information transmission.

Nine users of Advanced Bionics HiRes 90K cochlear implants and five users of a research implant with percutaneous connector (manufactured by Cochlear Ltd) participated in studies to compare temporal perception using tripolar and multipolar modes of stimulation, respectively, with that using monopolar stimulation. The ability to detect small differences in interpulse intervals within and across electrode positions was tested. It was hypothesized that the different spatio-temporal profile of neural activation in the different modes may lead to differences in within-channel temporal information transmission. It was further hypothesized that across-electrode interpulse interval differences would be more difficult to detect in the focussed modes due to fewer neurons receiving input from more than one electrode (thus creating more independence of within-channel temporal information). Neither of these hypotheses was supported by the data. A third hypothesis was that synchronous neural activation at different cochlear places may be important for complex listening tasks, and may therefore be perceptually distinguished from similar stimuli without synchronous electrode activation. This hypothesis was supported by preliminary data.

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S32: MONOPOLAR-COUPLED, ECTOPIC STIMULATION IN COCHLEAR IMPLANTS

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We have noted physiological evidence of an ectopic (out of place) site of electrical stimulation that may produce competing neural activity and may be associated with diminished cochlear implant (CI) performance.

We have examined subjects implanted with the Advanced Bionics (n=16) and Cochlear (n=17)CI systems. All subjects were studied using high-resolution CT imaging, intracochlear evoked potential (IEP) recording, electrode pitch discrimination, and measures of speech recognition. IEP recordings included measures of the spatial spread of neural responses obtained by holding the site and level of monopolar-coupled stimulation constant and incrementally changing the recording electrode site longitudinally along the array. In this case, variability in the N1 latencies of IEP responses is frequently observed, whereas normally no latency shift would be expected. When the latency shifts are observed, the response latencies become progressively shorter as the recording site is moved basally away from the apical stimulation site. Conversely, stimulating basally will frequently produce similar shifts toward shorter latencies as the recording electrode is moved apically.

To explain these paradoxical latency-shift observations, we hypothesize that the recorded responses are the net sum of neural responses from two anatomically-separate regions. One is the ganglion cell body region in the vicinity of the stimulating electrode, which responds with a long latency response traveling anterogradely out of the cochlea. The other is a site at the base of the cochlea where the VIII nerve exits via the *Tractusspiralisformaninosus* (TSF) into the internal auditory meatus (IAM). This secondary ectopic site is excited as monopolar-coupled current injected into the bone-enshrouded cochlea exits along the TSF, thus producing a local, short-latency neural response that travels both anterogradely toward the brain and retrogradely back into the cochlea. At the recording electrode within the cochlea, the recorded response is the net summation of responses from these two regions. By changing the position of the recording electrode, the relative contribution of the long- and short-latency sources is altered resulting in an apparent latency shift in the net response. Stimulation at the ganglion cell site is thought to be tonotopically-specific, whereas stimulation at the TSF is thought to be tonotopically-nonspecific, giving rise to conflicting, broad spectrum acoustic percepts. Some subjects with this phenomenon report competing noise and grunt/echo-like sounds which occur only when someone is speaking.

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S33: RELATIONSHIPS AMONG PERIPHERAL AND CENTRAL ELECTROPHYSIOLOGICAL MEASURES OF SPATIAL / SPECTRAL RESOLUTION AND SPEECH PERCEPTION IN COCHLEAR IMPLANT USERS

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Spectral resolution is an important feature of normal auditory processing and allows for discrimination among different sounds. Cochlear implant (CI) users demonstrate impaired spectral resolution abilities [e.g. Henry et al., 2005, JASA, 2, 1111-1121], which can be attributed at least in part to electrode interactions at the periphery (i.e. poor spatial resolution) [Jones et al., 2013, JASA, 133, 425-433]. The purpose of this study is to use electrophysiological measures to evaluate both spatial resolution (neural stimulation patterns of individual electrodes) and spectral resolution (discrimination among complex stimuli presented through the implant processor), and to determine how these measures relate to each other and to speech perception in CI users.

We performed extensive measures of peripheral spatial resolution (ECAP channel interaction functions) and evaluated central processing using the auditory change complex (ACC) in response to both spatial (electrode discrimination) and spectral (rippled noise) changes. We also assessed speech perception abilities (vowels in quiet and sentences in noise) within each subject. Three experimental 7-channel MAPs were created for each subject. The three MAPs were designed so that the potential for interaction among active electrodes varied: adjacent electrodes in MAP 1, every-other electrode in MAP 2, and every third electrode in MAP 3. Channel interaction functions were obtained for all activated electrodes, and spatial ACC responses were elicited by sequentially stimulating different pairs of activated electrodes. Subjects used the experimental MAPs to listen to spectrally complex stimuli (rippled noise and speech).

Collectively, the measures allow us to systematically evaluate auditory processing from the auditory nerve to the cortex, the processing of simple and complex stimuli, and ultimately, the relationship between electrophysiological measures of spatial / spectral information and speech perception. Preliminary results suggest a relationship among these measures. ECAP channel interaction functions were quantified as channel separation indices for electrode pairs [Hughes, 2008, JASA, 5, 2711-2714] and compared with spatial ACC amplitudes. Average channel separation indices were smallest for MAP 1 and largest for MAP 3, indicating that the experimental MAPs had the intended effects on spatial resolution. Spectral ACC amplitudes and speech perception scores tended to reflect spatial resolution measures; that is, they were poorest when subjects listened with MAP 1 and best with MAP 3. Correlation analyses will be performed once additional data are available.

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S34: VISUAL PROSTHESES: A PROGRESS REPORT TO THE CIAP COMMUNITY

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Retinitis pigmentosa (RP) is an inherited eye disease that leads to severe blindness following degenerative changes to the photoreceptors of the retina. RP affects over 1 million people worldwide and there is no drug based therapy that can stop this degenerative process. Importantly, large numbers of retinal ganglion cells are spared during the progression of RP, providing potential for a therapeutic solution via a neural prosthesis.

There are at least 23 research groups developing visual prostheses; the majority are developing retinal prostheses in order to directly stimulate retinal ganglion cells; other stimulation targets include the optic nerve and the visual cortex. I will spend the majority of this presentation discussing the development of retinal prostheses.

Clinical trials of retinal prostheses have demonstrated improved acuity for subjects with severe vision impairment that varies from bare light perception to a level where they can recognise shapes, navigate a room and read large letters. These devices consist of an electrode array surgically implanted on or near the retina in order to stimulate target neurons within the middle and inner retinal layers (e.g. bipolar and retinal ganglion cells). Epi-retinal electrode arrays, containing up to 60 platinum (Pt) electrodes, are attached to the surface of the retina using tacks. While this technique positions the electrodes close to their neural targets, the long-term mechanical stability of the electrode array and the potential requirement of routing a leadwire out of the vitreous cavity present challenges. Implants, containing up to 1500 titanium nitride electrodes, have been implanted sub-retinally in a small clinical trial. A sub-retinal placement of an electrode array is again positioning the electrodes close to their neural targets. Technical challenges include increased surgical difficulty in the safe placement of the prosthesis; the need to develop a very thin (sub-millimetre) electrode array/stimulator assembly and routing a leadwire out of the vitreous cavity. A third site for placement of an electrode array is within a tissue pocket created between the sclera and choroid. This site has a number of advantages including less traumatic surgery without the need to maintain a leadwire exiting the vitreous cavity; a larger electrode array covering a wider region of the retina; and long-term mechanical stability. The technical challenges for this approach are related to the electrodes being more distal to their neural targets resulting in increased thresholds and potentially lower selectivity compared with epi- and sub-retinal placements. We recently initiated a clinical trial using this approach.

While the clinical development of a visual prosthesis is clearly hampered by the complexity of the visual system, it is encouraging to note that the extent of research activity in this field has recently resulted in the first CE Mark and FDA regulatory approval of a retinal prosthesis as safe and effective for use in RP patients. The success of cochlear implants is an important factor in accelerating the development of visual prostheses.

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S35: CHALLENGE FOR TECHNOLOGICAL REGENERATION OF THE COCHLEAR SENSORY EPITHELIUM: DEVELOPMENT OF THE ARTIFICIAL COCHLEAR SENSORY EPITHELIUM USING PIEZOELECTRIC MATERIALS

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The cochlear sensory epithelium responds to sound-induced vibrations and converts these mechanical signals into neural signals, which stimulate the auditory primary neurons. Degeneration of the cochlear sensory epithelium is one of major causes for sensorineural hearing loss. We then attempted to fabricate an artificial cochlear sensory epithelium (ACSE), which functions as a sensor with the capability for acoustic/electric conversion without a battery, using micro-electro mechanical systems.

For this purpose, piezoelectric materials, which generate electricity in response to mechanical stress, were utilized. An ACSE is composed of a silicon frame and thin piezoelectric membrane. The first generation ACSE, although not life-sized, demonstrated a tonotopic map for frequencies of 6.6–19.8 kHz in air and generated maximum electrical output from an electrode placed at the site of maximum vibration. Based on this, we designed a miniaturized ACSE, which was implantable in the basal turn of a guinea pig cochlea. This second generation ASCE was able to generate electrical output ranging from 0.14 mV to 5.88 mV in response to sound stimuli at 100 dB SPL, at frequencies of 1–40 kHz in air. After implantation into a guinea pig cochlea, a piezoelectric membrane the second generation ACSE vibrated in response to sound stimuli similarly to the natural basilar membrane, and generated maximum voltage output of 29.3 μV, indicating a feasibility of a totally implantable auditory device using piezoelectric materials. However, there were many problems to be overcome before practical application. One of major problems was an increase of electrical output of the device. For this purpose, we then examined potential of other piezoelectric materials. Use of a new material and new design achieved 1000-time increase of electrical outputs, although not life-sized. In the near future, we will fabricate the third generation ACSE, and test its potential for hearing recovery after implantation.

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S36: PROGRESS TOWARD A MULTICHANNEL VESTIBULAR IMPLANT TO TREAT PROFOUND LOSS OF INNER EAR BALANCE SENSATION

Charley C. Della Santina, Chenkai Dai, Gene Y. Fridman, Russell Hayden, Natan S. Davidovics, Mehdi A. Rahman, Nicolas S. Valentin, Daniel Q. Sun, Joong Ho Ahn, Bryan Ward, Kristin Hageman, Americo A. Migliaccio, Zaven Kalayjian, Andreas Andreou, Francisco Tejada, Diana Mitchell, Kathleen E. Cullen and Abderrahmane Hedjoudje
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Bilateral profound loss of vestibular sensation is disabling. Affected individuals suffer illusory drift of visual fields during head movement, chronic disequilibrium and postural instability caused by failure of vestibulo-ocular and vestibulo-spinal reflexes. Data from the Balance and Dizziness Supplement to the 2008 United States National Health Interview Survey reveal that 81/100K U.S. adults suffer a constellation of symptoms consistent with chronic, severe or profound bilateral loss of vestibular sensation. This extrapolates to ~500,000 adults in the U.S. & Europe, or ~3 million worldwide. Those reporting this symptom complex often stopped driving due to their symptoms and incurred a 24-fold increase in fall risk in comparison to the nationwide average. There is no adequate treatment for individuals disabled by bilateral vestibular loss despite optimal medical and rehabilitative intervention. An implantable prosthesis that accurately emulates normal sensory transduction of head rotation could significantly improve quality of life for these individuals. Progress toward such a device has accelerated over the past decade, and clinical feasibility trials of similar devices designed to deliver transient stimulation of the labyrinth have yielded promising results in humans. Achieving sufficient stimulation selectivity and intensity to accurately and chronically/continuously encode the full range of natural 3-dimensional head movements remains a key challenge. This lecture will provide an overview and update on prosthetic vestibular stimulation research at the Johns Hopkins Vestibular NeuroEngineering Lab, including the neurophysiologic basis of current prosthesis development efforts; comparison of different stimulation and encoding strategies; impact of vestibular implantation on hearing; directional plasticity and adaptive responses to chronic multichannel stimulation; infrared laser stimulation; novel approaches to inhibiting undesired neuronal activity; and progress toward a system for clinical application.

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S37: IMPLANTABLE HEARING SYSTEMS – RATIONALE, CONCEPTS, TECHNOLOGY, CLINICAL RESULTS AND PERSPECTIVES

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Auditory implants have enlarged the treatment portfolio for all types of hearing loss. They include systems using electrical stimulation, e.g. cochlear implants, and/or mechanical stimulation of the auditory system. This presentation will focus on the latter group.

Mechanical stimulation is used to functionally restore impaired mechanical hearing processes of the outer, middle and inner ear. The electro-magnetic or piezo-electric actuators are coupled on the skull, the ossicles of the middle ear or the inner ear at the round window membrane. Direct cochlear mechanical stimulation is achieved by perilymph coupling through the footplate with increasing sound pressure and broadened frequency transfer into the inner ear.

The devices in clinical use can be classified as per- or transcutaneous partially or fully implantable systems. They are used to treat patients with pure conductive, mixed or sensorineural hearing loss with sufficient cochlear reserve.

The paper will give an overview of the different systems in clinical use (bone-anchored hearing aids, middle ear transducers and direct acoustic cochlear stimulators), key technical parameters, medical indications, surgical aspects, audiological results and complications. The last part will focus on the combined electro-mechanical stimulation for hybrid systems and future developments including the multimodal inner ear probe as a theranostic device to functionally map the inner ear.

MONDAY POSTER ABSTRACTS

M1: MIXED STIMULATION RATES FOR ENCODING TEMPORAL CUES IN BILATERAL COCHLEAR IMPLANTS

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Bilateral cochlear implants (CIs) provide clear benefits to recepients, but there is a significant gap in performance on spatial hearing tasks relative to performance of normal-hearing (NH) listeners. Much of the ability of NH listeners to localize sounds in the horizontal plane depends upon being able to extract and utilize the interaural time differences (ITDs) in the “temporal fine structure” (TFS) of sound between the two ears at frequencies below 1.5 kHz. Most contemporary CIs stimulate the auditory nerve with constant-rate, high-rate electric pulse trains and discard the TFS. This strategy may allow for adequate envelope ITD representation, but the loss of TFS may account for many of the deficits in binaural hearing abilities compared to NH. Though CI listeners can use the timing cues in low-rate pulse trains for ITD discrimination and monaural rate discrimination, better speech recognition is generally achieved with high-rate stimulation. The present study investigated this dichotomy. We propose a set of processing parameters which allows for sufficient speech envelope representation as well as pulse timing utilization for ITDs.

Three tasks were used to systematically examine the effects of channel pulse rate and pulse timing on ITD discrimination, ITD lateralization, and recognition of speech in quiet by CI listeners. Stimuli consisted of speech tokens processed at low (~140pps), high (~1200pps), and mixed (~140pps on 4 apical electrodes/~1200pps on 4 basal electrodes) electrical stimulation rates. Stimulation was presented using research processors that enable pulses in the two ears to be synchronized across select pairs of electrodes. We used 8 pitch-matched electrode pairs. For each rate combination, two strategies were tested; the first strategy preserves a representation of acoustic TFS information in the pulse timing, and the second strategy (Continuous Interleaved Sampling) does not. Results show that CI listeners were able to use low-rate pulse timing cues derived from acoustic TFS for ITD discrimination and lateralization of multi-channel speech stimuli. When only high rates were present, CI listeners were able to discriminate, but not lateralize, with ITDs, regardless of processing strategy, indicating the use of envelope ITDs to perform the task. Speech understanding was best whenever high stimulation rates were present. These results suggest that a mixed-rate paradigm may be optimal in providing both the desired level of speech understanding due to high stimulation rates on some electrodes, and also ITD sensitivity due to low-rate timed pulses on others. This work may have implications for the design of new speech processing strategies that maximize binaural hearing.

Work is supported by NIH-NIDCD (R01-DC003083, Litovsky).

M2: BINAURAL SENSITIVITY OF BILATERAL COCHLEAR IMPLANTED PATIENTS TO AMPLITUDE MODULATED STIMULATION PRESENTED ON MULTIPLE ELECTRODES

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Current cochlear implant speech processing strategies use high pulse rate s (~1000pps) on multiple electrodes to achieve good speech understanding. However, when using these strategies, sound localization performance is poor, most likely due to poor encoding of interaural time differences (ITDs). Studies using direct electrical stimulation have demonstrated that many bilateral cochlear implant (BiCI) users show varying degrees of sensitivity to interaural time differences (ITDs) when stimulated with low rate (~100pps) constant amplitude or with high rate (~1000pps) amplitude modulated pulse trains on single interaural pairs of pitch-matched electrodes at different cochlear locations. As a first step towards developing new coding strategies which maintain good speech understanding while also providing ITDs, there is a need to understand ITD sensitivity in multiple electrode stimulation. For example, is a patient's sensitivity to ITDs on multiple electrodes limited by the electrode pair that produces the lowest sensitivity or is performance directed by the pair with the best sensitivity? In addition, should different cochlear places be stimulated at various rates, in efforts to increase ITDs sensitivity?

In an experiment using synchronized research processors, ITD thresholds were measured using a two-interval, two-alternative force choice task for five pitch-matched pairs of electrodes spanning the base-mid-apex length of the array. Pairs were tested either individually, three at a time, or all five together. In the multi-electrode conditions, the same ITD was applied to all pairs. Various configurations of 3-electrode pairs were tested: (1) basal; three most basal pairs (2) apical; three most apical pairs (3) spread; base-mid-apex pairs (4) best; three pairs with the lowest ITD thresholds when measured individually. ITD sensitivity was measured for low rate (100pps) constant amplitude and high rate (1000pps) amplitude modulated (100 Hz) pulse trains.

Results show that there was no particular cochlear place with the best ITD sensitivity across subjects for either unmodulated or modulated stimuli in any electrode configuration (single or multiple). Furthermore, the spread multi-electrode (base-mid-apex) configuration often resulted in better ITD sensitivity than either electrode grouping in one place (basal, apical) or using the best configuration. However for the amplitude modulated stimuli, the best configuration of 3-electrode pairs generally produced the best ITD sensitivity across subjects. The 5-electrode configuration was either about the same or better than the most ITD sensitive single pair for both modulated and unmodulated stimuli. These data indicate that ITD sensitivity to multi-electrode stimulation is comparable to thresholds measured for single electrode pairs. Outcomes from this study have implications for developing speech processing strategies which better incorporates or enhances ITD information for BICI users by increasing a signal's modulation depth.

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M3: LISTENING IN A COCKTAIL PARTY:EVIDENCE FOR BINAURAL PROCESSING BY BILATERAL CI AND HEARING PRESERVATION LISTENERS

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Two groups of cochlear implant (CI) listeners were tested for sound source localization and for speech recognition in complex listening environments. One group (n=11) wore bilateral CIs and, potentially, had access to interaural level difference (ILD) cues, but not interaural time difference (ITD) cues. The second group (n=8) wore a single CI and had low-frequency acoustic hearing in both the ear contralateral to the CI and in the implanted ear. These 'hearing preservation' listeners, potentially, had access to ITD cues but not to ILD cues. At issue was the value of the two types of information about sound sources, ITDs and ILDs, for localization and for speech perception when speech and noise sources were separated in space.

For Experiment 1, normal hearing (NH) listeners and the two groups of CI listeners were tested for sound source localization using a 13 loudspeaker array. For the NH listeners, the mean RMS error for localization was 7 degrees, for the bilateral CI listeners, 20 degrees, and for the hearing preservation listeners, 23 degrees. The scores for the two CI groups did not differ significantly. Thus, both CI groups showed equivalent, but poorer than normal, localization. This outcome using the filtered noise bands for the normal hearing listeners, suggests ILD and ITD cues can support equivalent levels of localization.

For Experiment 2, the two groups of CI listeners were tested for speech recognition in noise when the noise sources and targets were spatially separated in a simulated 'cocktail party' environment. At issue was whether either CI group would show evidence of binaural hearing, i.e., better performance when the noise and targets were separated in space, indicating the ability to 'unmask' speech in noise. Binaural advantage compared (i) the better CI ear to bilateral CIs allowing access to ILD cues when the poorer CI was added, and (ii) bimodal to the combined condition (bilateral acoustic hearing + CI) when the preserved hearing was added allowing access to ITD cues. Both CI groups showed a significant binaural advantage (a combination of squelch and summation) of 18 percentage points. Results show that either cue "ITD or ILD" allows similar levels of performance when listening in complex noise environments.

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M4: WHICH CHILDREN SHOULD BE CONSIDERED CANDIDATES FOR BILATERAL COCHLEAR IMPLANTATION?

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It is increasingly common to offer cochlear implants to children with residual hearing. This trend prompts a question: what is the minimum degree of hearing impairment at which a child is likely to gain more benefit from implants than from hearing aids? To help answer this question, we are conducting a longitudinal comparison of outcomes for children with bilateral cochlear implants and children with bilateral acoustic hearing aids. The aim is to define a criterion of candidacy for bilateral implantation.

Twenty-seven of the participants use bilateral cochlear implants and have severe-to-profound hearing impairment. Forty-four of the participants use bilateral digital hearing aids and have hearing impairment ranging from mild to profound. Children are assessed twice with an interval of a year between assessments. All children have completed the first assessment (mean age 4.3 years, range 3.0 to 6.3 years). To date, 47 participants have completed the second assessment (mean age 5.4 years, range 4.0 to 7.2 years). At both assessments, participants attempted tests of 1) language skill; 2) sound-source localization; 3) speech perception in quiet, in random noise, and in babble.

Data from the first and second assessments were analyzed separately. For each outcome measure, a regression function was used to characterize the relationship between unaided hearing level and performance for children with hearing aids. The distribution of scores for implanted children was used to calculate, for a newly-diagnosed child with a known hearing level, the odds of better performance with implants than with hearing aids. A criterion of candidacy can be defined as the most advantageous hearing level for which the odds of better performance with implants exceed an acceptable ratio, such as 4 to 1.

At the first assessment, the four-frequency pure-tone average hearing level (PTA) associated with odds of 4 to 1 was 84 dB HL for the measure of speech perception in quiet. For the remaining outcome measures, odds of 4 to 1 were associated with PTAs greater than 100 dB HL. However, results at PTAs above 90 dB HL should be interpreted with caution. This is because only four participants with hearing aids had a PTA greater than 90 dB HL, so there are few data points to inform the regression line above this level. Based on the results of the first assessment, children with PTA greater than 84 dB HL should be considered candidates for bilateral cochlear implantation. This criterion of candidacy is very similar to current practice in the UK.

Interim data from the second assessment indicate that the four-frequency PTA associated with odds of 4 to 1 ranged from 76 to over 100 dB HL, depending on the outcome measure. At the conference, we will present additional data from the second assessment and we will assess whether the results are stable over time.

Supported by Action on Hearing Loss.

M5: PITCH RANKING IN BILATERAL ASYMMETRIC COCHLEAR IMPLANT ELECTRODE ARRAYS

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Introduction: Pitch perception in cochlear implant users is mainly dependent on the place of stimulation. In bilateral cochlea implant users, electrode array design differences between ears will induce individual places of excitation. Consequently, a certain electrode number corresponding to a specific frequency band will perhaps elicit a different pitch in comparison to the same electrode number in the opposite ear. Even with identical electrode arrays, insertion depth differences will cause pitch height differences. Here, we report on (1) pitch ranking results in bilateral implanted patients with different electrode arrays in each ear, (2) cases with equal electrode design, but different electrode insertion depth as a consequence of unilateral electrode migration and (3) the outcome of the attempt to reinsert the migrated electrode array in terms of pitch ranking results will be presented.

Material and Methods: Loudness balanced single electrode stimuli (pulse trains, 500 ms duration) were presented in sequence either unilaterally or between ears. After presentation of the stimuli on tested electrodes, the task of the subject was to report whether the pitch of the second stimulus was lower, equal or higher compared to the first stimulus. In order to assess individual electrode discrimination, unilateral pitch ranking was assessed prior to the bilateral ranking test. Pitch ranking was studied (1) in one bilateral subject implanted with two MED-EL (Innsbruck) devices, each carrying 12 double electrodes but different electrode lengths (24 mm left, 15 mm right), (2) two subjects who experienced a unilateral migration of the electrode array, probably as a result of trauma (MED-EL Flex28/FlexSoft) and, (3) after successful re-insertion surgery. The results were analyzed referring to Reiss et al [Ear and Hearing, 32, 536-540].

Results: Unilateral pitch ranking was perfect in all tested ears. Bilateral pitch ranking results for the different cases are as follows (1) Different electrode arrays: Bilateral pitch ranking showed an adapted pitch perception (electrode mapping from the short to medium electrode): E3 to E1, E8 to E5/E6 and E12 to E11. (2) Electrode migration: ranking shifted up to higher electrodes, mean shift two electrodes, in line with the corresponding x-ray (cochlear view). (3) Reinsertion of migrated electrode array: The retest of pitch ranking showed a normalized pitch perception without pitch shift.

Conclusion: The present results demonstrate the benefit obtained by successful re-insertion surgery to a normalized bilateral pitch perception for the two subjects after electrode migration. The subject with different electrode lengths showed a partial adaption to the center frequencies of the analytic frequency filter of the speech processor.

M6: AUDIO-VISUAL SPEECH INTELLIGIBILITY BENEFITS WITH BILATERAL COCHLEAR IMPLANTS WHEN TALKER LOCATION VARIES DYNAMICALLY

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In noisy social settings talker location often varies dynamically and the ability to readily turn towards a talker of interest provides gains in intelligibility and ease of listening through the availability of visual cues such as lip-reading. Orientation towards the target direction also ensures good signal-to-noise ratios (SNRs) at the ear(s), whereas particularly for unilateral listening, incorrect orientation can produce much poorer SNRs. To obtain an objective performance measure under such circumstances bilateral implantees were assessed for speech intelligibility in noise using a novel spatial audio-visual test paradigm. Target audio-visual sentences were randomly presented from one of four locations spanning 180º in the frontal arc, and randomly drawn from two female and two male talkers. At the same time visual-only distractors depicting the three non-target talkers were presented at non-target locations. Target sentences were preceded by an audio-only cueing sentence spoken by the same talker, at the same SNR, and from the same direction as the target. An audio-only noise field was activated before the cueing sentence and switched off after completion of the target sentence. The noise comprised continuous discourse from an additional 8 talkers, each presented from one of 8 individual loudspeakers spanning 360º, combined with a spatial rendering of the reverberation produced by that noise field as it would be experienced in a large cafeteria. Intelligibility was assessed using an adaptive procedure to estimate the SNR producing 50% correct performance (SRT50). The test was repeated for each ear alone and bilaterally, and both with and without visual cues.

In the audio-visual condition, results showed large bilateral improvements ranging between 5dB and 15dB in terms of SRT50 reductions compared to the better outcome with either ear alone. While bilateral benefits were largest for audio-visual presentation, smaller benefits were also seen for the audio-only condition, suggesting that both improved SNR and visual cues contribute. In the binaural condition, the inclusion of visual cues at the target location led to an average SRT50 improvement of about 5dB over the audio-only condition. For the monaural condition no such benefit arose, presumably because the visual cues were not seen. These outcomes illustrate that when target location varies dynamically, the speech intelligibility gain in noise associated with bilateral device use can be much larger than indicated by previous objective measures that have used static spatial configurations.

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M7: MEASURING THE BILATERAL ADVANTAGE IN COCHLEAR IMPLANTEES USING A BLIND MODEL

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Several studies have shown that bilateral, compared to unilateral, cochlear implantation improves the identification of speech-in-noise. This bilateral advantage has recently been studied by Culling et al. [Ear & Hearing 33 (6), 673-682 (2012)], where a model was described able to make predictions of the improvement in signal-to-noise ratio (SNR) for various spatial configurations, as a result of bilateral implantation. Whilst achieving very high correlations with actual CI speech reception thresholds (SRTs), the above model has the practical limitation of requiring information not directly available to the listener, such as the room impulse response. In this study, we develop and evaluate a blind measure to predict the bilateral advantage to speech intelligibility based on cochlear implant auditory-inspired insights.

Such a model, termed BiSIMCI, combines an equalization-cancellation stage together with a modulation frequency estimation stage. Cochlear implant specific hearing constrains, such as the limited interaural phase difference sensitivity or the absence of fine-structure, are taken into account by the BiSIMCI model.

Simulated and actual subjective data from cochlear implant users were used to validate the proposed model. The actual CI subjective data consisted of SRTs in anechoic rooms, and with a total of 28 different target/interferer spatial configurations. The simulated CI subjective data was obtained by running the non-blind algorithm by Culling et al. across 109 different target/interferer conditions from two environments, one anechoic and one highly reverberant room (RT60=0.89s).

The experimental results indicate that the proposed blind model provides reliable predictions when compared with both actual and simulated cochlear implant SRT data for spatially distributed interferers. An average correlation of 0.94 was reported for these conditions, and an average correlation of 0.97 was obtained when some a priori information was added to the model. Further investigations will be directed towards solving some of the model limitations.

M8: RELATION BETWEEN INTERAURAL ENVELOPE DELAY SENSITIVITY AND SPEECH ENVELOPE STATISTICS IN ELECTRIC STIMULATION

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In normal hearing, interaural time differences (ITDs) at low frequencies are considered important for sound localization and spatial speech unmasking in the lateral dimension. It has been shown that highly salient lateralization cues can also be provided by encoding the ITDs in appropriately shaped envelopes of high-frequency sounds.

In this study, we measured the sensitivity of seven bilateral cochlear-implant (CI) listeners to envelope ITD. In particular, we were interested in the independent perceptual contribution of different features of the ongoing envelope modulation on left/right discrimination thresholds. The binaural stimulus consisted of a 1515-pulse/s pulse train modulated with a 27-Hz trapezoidal envelope, presented at an interaurally pitch-matched electrode pair. The modulation trough level corresponded to the listener-specific absolute threshold for an unmodulated, 1515-pulse/s pulse train. The slopes were linear on a linear current scale. We systematically varied the off time (the silent interval in each modulation cycle) and the slope of the trapezoid flanks. As a special case of modulation with infinitely steep slopes and maximum possible off time we also tested a 27-pulse/s pulse train. The peak levels were adjusted to obtain constant loudness and centralized auditory images across all conditions.

Increasing the off time up to 20 ms improved sensitivity. While increasing the slope did not show a systematic effect, the effect can be anticipated for shallower slopes than those tested. Our restriction on the modulation depth prevented testing such shallow slopes. The sensitivity for the unmodulated pulse train was considerably higher than that for the best-performance condition with trapezoidal modulation, which we attribute to the larger peak level.

The results were compared with the statistics of envelope parameters extracted from German and English speech recordings processed by a simulation of a typical envelope-based CI strategy. For the off-time analysis, we used a definition based on the silent interval close to the envelope peak level [Dietz et al., 2013, J. Acoust. Soc. Am. 133, 1-4]. The comparison showed that while CI-processed speech already contains some potentially perceivable envelope ITD cues, there is room for enhancing those cues by signal processing, particularly with respect to the off time.

Providing more salient envelope ITD cues might contribute to sound localization and source segregation in situations where pulse-ITD cues are ambiguous, disrupted by room reverberation, or implicitly not encoded.

M9: IMPROVING LOCALIZATION BY PRESERVATION OF ILD CUES

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Normal-hearing listeners use binaural difference cues, interaural level differences (ILDs) and interaural time differences (ITDs), to localize sounds and improve speech understanding in spatially separated noise. Bilateral cochlear implants have proven advantageous over unilateral devices, but there are still opportunities to improve access to binaural cues with two cochlear implants. In this presentation, we discuss approaches to better preserve ILD cues.

Bilateral cochlear implant recipients using commercial strategies primarily rely on ILD cues to localize sounds in the horizontal plane. In order to better preserve ILD cues, it is important to address possible distortions from the independent signal processing in the two ears. One possible source of distortions is the automatic gain control (AGC). For example, when loud sounds are at one side of the head, the AGC can compress the signal in the near ear in a time-varying and different way than the far ear, thus distorting ILDs. We have examined synchronizing gain information across sides as well as raising the level at which the AGC is triggered (i.e., the AGC knee point).

The localization ability of Nucleus bilateral cochlear implant recipients was studied with 3 variations in signal processing: standard AGC, bilaterally-linked AGC, and raised AGC knee point. The RMS error was measured in these conditions using the Iowa Environmental Sounds with an 8 speaker array (spaced 20 degrees apart from -70 to +70).

For the extreme lateral locations, an improvement in RMS error of 13 degrees ($p<0.05$; $N=9$) was found with linking the AGCs. At the level tested (75 dB SPL), similar and significant improvements were found by raising the AGC knee point. Both approaches resulted in improved localization of lateral, loud sounds, while not having a negative impact on perception of other sounds (either at more central locations or lower levels).

CJL, LR, WSP, ZMS are employed by Cochlear Ltd.

M10: THE PRECEDENCE EFFECT: INSIGHTS FROM ELECTRIC HEARING

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Normal-hearing listeners localize sound sources in everyday environments on the basis of the directional cues carried by the first-arriving sound rather than spurious cues associated with later-arriving echoes and reverberation, a phenomenon known as the precedence effect. Limited data from bilateral cochlear implant (CI) users suggest the effect is weak or nonexistent in individuals wearing their clinical processors, in turn suggesting that hypersensitivity to spurious spatial acoustic information may contribute to listening difficulties in complex acoustic scenes. The present study thus aimed to characterize the precedence effect in bilateral CI users given bilaterally synchronized stimulation. In addition to providing insight on an ecologically important perceptual phenomenon in a growing patient population, the study enabled evaluation of competing hypotheses concerning the physiologic basis of the precedence effect: One hypothesis holds that the effect is mediated by central mechanisms (e.g., synaptic inhibition), while the other holds that it is mediated by peripheral mechanisms (e.g., cochlear resonances or auditory nerve adaptation). Under the central hypothesis, bilateral CI users with good binaural sensitivity and ostensibly preserved central auditory function should experience a relatively normal precedence effect; under the peripheral hypothesis, they should experience little or no precedence effect.

Adult bilateral CI users with demonstrated sensitivity to interaural time differences (ITDs) were tested in a “lead-lag” lateralization paradigm: A biphasic pulse (the “lead” or direct sound) was delivered bilaterally to pairs of pitch-matched electrodes via synchronized research processors with a 500-μs right- or left-favoring ITD, followed 1-64 ms later by a second bilateral pulse (the “lag” or simulated reflection) with an opposing 500-μs ITD. Subjects used a touch-sensitive display to indicate on each trial (1) whether one or two locations were perceived and (2) the location perceived. In the event two locations were perceived, subjects were further instructed to indicate the left-most location perceived. The task thus simultaneously assessed lead-lag fusion and the extent to which lateralization was influenced by lead versus lag ITD cues.

On average, subjects exhibited a weaker precedence effect than normal-hearing subjects tested in similar paradigms. Nonetheless, the effect was clearly present in a majority of subjects, refuting recent suggestions that the precedence effect is entirely attributable to cochlear processes. In an additional condition, the amplitude of the lag stimulus was systematically attenuated to mimic adaptation observed in a model auditory nerve’s response to an acoustic lead-lag signal. Attenuation of the lag proportional to normal auditory nerve adaptation moderately increased the precedence effect. Data are thus consistent with contributions of both central and peripheral mechanisms. Importantly, data collectively suggest that the precedence effect can be restored in bilateral CI users with bilaterally synchronized stimulation.

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M11: NEURAL CODING OF INTERAURAL LEVEL DIFFERENCES WITH BILATERAL COCHLEAR IMPLANTS

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Although interaural level differences (ILD) are by far the most potent sound localization cues presently available to bilateral cochlear implant (CI) users, virtually nothing is known about how the brain encodes ILDs of electrical stimuli. To fill this gap in knowledge, we characterized ILD sensitivity of single neurons in the inferior colliculus using two animal models of bilateral CI: acutely-deafened, anesthetized cats and awake rabbits chronically implanted in both ears for up to 6 months.

Stimuli were trains of biphasic current pulses presented at rates <100 pulses/s. Current amplitude was independently varied in each ear starting just below threshold and spanning the entire dynamic range. Four sets of curves were extracted from these data: 1) firing rate vs. contralateral level (LC) at different ipsilateral levels (LI), 2) rate vs. LI at different LC, 3) rate vs. ILD at different mean binaural levels (MBL) and 4) rate vs. MBL at different ILDs. For each neuron, we determined which of the four level variables (LC, LI, ILD, MBL) was most robustly coded by finding which of the four set of rate curves was most invariant to variations in the cross-cutting parameter (LI, LC, MBL and ILD, respectively).

Over half the neurons (60-70%) in both preparations were dominated by contralateral excitation and hence best coded LC. In awake rabbit, ILD-coding neurons were the next most common (14%). These neurons showed robust, monotonic rate-ILD curves over a range of MBLs, providing a nearly unambiguous code for ILD. The proportion of ILD-coding neurons was lower (8%) in anesthetized cat, where MBL-coding neurons were the second most common (27%). These neurons were excited by stimulation of either ear; their rate-ILD curves were nonmonotonic and highly dependent on MBL, thereby ambiguously coding ILD. Few neurons in either preparation best coded LI (6% in cats, 4% in rabbits). Several factors may contribute to the different proportions of coding types in acutely-deafened, anesthetized cats vs. chronically-implanted awake rabbits, including effects of anesthesia, species differences, duration of deafness, and sampling biases.

Overall, sensitivity to binaural level with electric stimulation qualitatively resembles that previously observed in normal hearing animals. Our four coding types (LC, LI, ILD and MBL) roughly correspond to the traditional EO, OE, EI, and EE classifications, respectively, but with an explicit emphasis on the robustness of the neural code. In both acoustic and electric hearing, ILD could be coded both in the firing rates of individual ILD-coding neurons and in the pattern of activity across a population of MBL-coding neurons differing in ipsilateral and contralateral thresholds. The similarity in ILD coding between acoustic and electric hearing is consistent with the good perceptual ILD sensitivity observed in CI listeners.

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M12: PRE- AND POST-OPERATIVE BINAURAL UNMASKING FOR BIMODAL COCHLEAR-IMPLANT LISTENERS

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Cochlear implants (CIs) are increasingly recommended to individuals with bilateral acoustic hearing sensitivity. However, important binaural benefits for speech perception in noise could be sacrificed as a result. Although new electrode designs and surgical approaches show promise for hearing preservation, today most CI recipients are expected to lose acoustic hearing in the ear implanted. While overall speech perception is usually improved due to the benefits of combined electro-acoustic hearing, bimodal listeners generally cannot take advantage of binaural unmasking to aid speech reception in noise. This study examined the tradeoff between the benefits of a CI and the potential loss of binaural unmasking for individuals who become bimodal CI users.

It is difficult to evaluate whether bilateral acoustic hearing can facilitate binaural unmasking in CI candidates because pre-operative speech-reception in noise, where binaural unmasking is most likely to occur, is generally too poor to reliably measure. This study employed an audiovisual speech-perception test to increase performance to measureable levels. Binaural Masking Intelligibility Differences (BMIDs) were measured prior to cochlear implantation (N=11) and at least six months post-activation (N=9 of 11 to date). IEEE sentences were presented in speech-shaped noise at signal-to-noise ratios between -10 and +15 dB. The noise was always correlated between the two ears. The speech was either correlated (NoSo) or inversely correlated (NoSpi). Signals were delivered via headphones at ~95 dB SPL (speech level) to the unaided ear(s) and via direct connection to the CI processor at MCL. BMIDs were calculated as the dB difference between each subject's NoSo and NoSpi psychometric functions.

Five of 11 subjects showed a significant pre-operative BMID (range: 2.0-7.3 dB). Four of these five subjects have been tested post-operatively. While none showed a significant BMID, three of the four subjects improved in the NoSo condition, yielding performance equal to or better than their pre-CI NoSpi performance. Five of the six subjects without a pre-operative BMID have been tested post-operatively. None showed a BMID, but all had performance equal to or better than their best pre-CI performance.

The results demonstrate that some individuals with bilateral acoustic hearing who receive a CI may experience reduced binaural unmasking when the source of interest is spatially separated from the background noise. However, the loss of this binaural advantage was in most cases compensated for or exceeded by an overall benefit to speech perception provided by the CI. Evidence of a pre-operative BMID nevertheless suggests that an individual might benefit from hearing-preservation surgery to preserve acoustic binaural unmasking. This test of binaural audiovisual speech-perception in noise could serve as a diagnostic tool to identify candidates likely to benefit from such a procedure.

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M13: BINAURAL SPEECH-STREAM SEGREGATION FOR SINGLE-SIDED DEAF AND BILATERAL COCHLEAR IMPLANTEES

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Binaural listening provides a tremendous advantage for normal-hearing (NH) listeners in complex auditory environments. Having two ears allows the listener to take advantage of head-shadow effects by selectively attending to the ear that provides the best signal-to-noise ratio (SNR) (i.e., the “better-ear” advantage) and provides access to binaural difference cues that facilitate the perceptual segregation of spatially-separated sources. Unfortunately, most research on listeners with cochlear implants (CIs) has shown that their binaural listening abilities can be predicted almost entirely from better-ear SNR advantages, suggesting that they receive little or no benefit from the binaural difference cues that can significantly enhance stream segregation for NH listeners.

We present evidence suggesting that CI users can, in fact, take advantage of binaural difference cues to enhance speech-source segregation in certain situations. We evaluated the binaural listening abilities of two types of CI users: single-sided deaf (SSD) CI users, with one NH and one CI ear, and bilateral CI users. Listeners completed a task based on the Coordinate Response Measure, requiring them to segregate a target talker from one or two masking talkers in the first (NH or CI) ear. The second (CI) ear was presented with silence or with a mixture containing only the maskers. This allowed us to test whether listeners could binaurally integrate the masker signals to better hear the monaurally-presented target.

Preliminary results from three (of six planned) SSD listeners show that presenting the maskers to the CI ear improved performance, but only in conditions where the target talker was masked by one or two interferers of the same gender as the target. Preliminary results from two (of eight planned) bilateral CI users show that presenting the maskers to the second CI improved performance in conditions involving one interferer, regardless of gender. This difference likely reflects the fact that the SSD users were able to use pitch cues to segregate opposite-gender voices in the target ear, while bilateral CI users were not. These results were confirmed and extended with vocoder experiments using a word-identification test and unrelated maskers.

Overall, these results suggest that a second auditory input via CI can provide binaural cues to facilitate the perceptual segregation of competing speech streams presented to the first (NH or CI) ear, but only for those interfering-talker situations where monaural cues fail to promote stream segregation.

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M14: BINAURAL UNMASKING WITH TEMPORAL FINE STRUCTURE IN COCHLEAR IMPLANT LISTENERS

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Binaural hearing typically provides improved speech perception and signal detection in noisy environments when there is spatial separation between sources. However, this binaural advantage is severely limited or absent for listeners with bilateral cochlear implants (BiCIs). While BiCI listeners have shown binaural benefits for signal detection with direct electrical stimulation, thresholds in dichotic conditions have been higher than those observed in normally hearing listeners. To date, the stimuli used for examining binaural unmasking in BiCI listeners have contained binaural information in the temporal envelope while the timing of the carrier pulse train has contained no usable information. There is evidence that BiCI listeners are sensitive to interaural time differences in bilaterally-synchronized aperiodic pulse trains for nominal pulse rates as high as 1600 pulses per second (pps) [Laback and Majdak, 2008, ProcNatlAcadSci USA, 105, 814-817]. The present study examined whether inclusion of temporal fine structure information in the carrier (i.e., presenting aperiodic pulses) would improve binaural unmasking for BiCI listeners due to greater access to interaural timing information.

Signal detection thresholds were measured in adults with BiCIs. All listeners had measurable interaural time difference thresholds for 100-pps constant amplitude pulse trains. Stimuli were constructed from 400-ms noise bands with center frequencies of 500 or 750 Hz and bandwidths of 50 or 125 Hz. Target stimuli had a 300-ms tone embedded in the noise. The noise was interaurallyhomophasic (N0) while the tone was either homophasic (S0) or antiphasic (Spi). Temporal envelopes were presented with either a synchronized steady-rate carrier at 500, 750, or 1000 pps or a carrier with pulses timed to the positive peaks in the fine structure. Stimuli were presented to single pairs of pitch-matched electrodes.

Results from 4 listeners (8 to be tested in total) show lower N0Spi thresholds than N0S0 thresholds by about 12 dB in all conditions indicating listeners were sensitive to interaural information in the temporal envelope. N0Spi thresholds did not improve from the inclusion of fine structure information for these listeners indicating that they were not benefiting from the provision of interaural information in the carrier. One listener showed improved thresholds for N0S0 stimuli with the inclusion of fine structure at 750 pps compared to N0S0 stimuli without fine structure at 1000 pps for both bandwidths, suggesting fine structure allowed the listener to use temporal pitch cues. The results thus far suggest that interaural timing information in the carrier pulses does not improve thresholds beyond what BiCI listeners achieve with interaural information in the temporal envelope.

Support provided by NIH-NIDCD R01003083 (Litovsky) and P30 HD03352 (Core Grant of the Waisman Center).

M15: WITHIN-SUBJECT COMPARISONS OF ELECTROPHYSIOLOGIC AND PSYCHOPHYSICAL MEASURES OF SPECTRAL RIPPLE DENSITY DISCRIMINATION IN CI USERS WITH SINGLE SIDED DEAFNESS.

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Cochlear implant (CI) candidacy has changed dramatically over the past three decades. With advancements in technology and a greater understanding of the long-term physiologic effects of cochlear implants, candidacy requirements have relaxed and individuals with greater amounts of residual hearing are receiving CIs. In this report we focus on an emerging population of CI users: individuals with single-sided deafness (SSD). Results from previous experiments using complex, spectrally modulated noise to assess the spectral resolution abilities of both normal hearing and hearing impaired listeners have shown promising correlations to speech perception (Supin et al., 1998; Henry and Turner, 2003; Henry et al., 2005; Litvak et al., 2007; Won et al., 2011). A limitation of these studies is that they rely on cross-groups comparisons. This presentation will describe within-subject comparisons of cortical and behavioral responses to spectrally complex stimuli by listeners who have normal (or near normal) hearing in one ear and use a CI in their impaired ear.

Behavioral and physiologic responses will be collected from 3-5 participants. The stimuli used for both the psychoacoustic and evoked potential studies will be complex, modulated noise stimuli with varying rectified spectral peak (ripple) densities ranging from 0.25-18.0 ripples per octave. Each ear will be assessed independently. Stimuli will be presented to the normal hearing ear via insert earphone with the CI turned off. The implanted ear will be tested with the signal routed into the processor via the direct audio input jack on the subject's speech processor. In the behavioral paradigm, spectrally modulated noise will be presented in a three-interval, three alternative forced choice (3I-3AFC) procedure. One interval will contain a stimulus where the position of the peaks and valleys in the spectral ripples has been inverted. The subject's task is to identify which interval contains the "different" stimulus. Ripple density will be systematically varied until a ripple density discrimination threshold is identified. During electrophysiologic testing, cortical onset and change responses will be elicited using the same ripple stimuli as described for the psychophysical testing.

We anticipate finding differences between the normal hearing and CI ears for both the behavioral and electrophysiologic measures of ripple density discrimination. We also expect the electrophysiologic measures to be less sensitive but correlated with the behavioral measures. This type of direct, within-subject comparison may help us better understand how spectrally complex acoustic information is processed through the auditory system of CI users. It is hoped that these comparisons may lead in turn to the development of more sophisticated processing strategies.

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M16: BILATERAL CHANGES OF SPONTANEOUS ACTIVITY WITHIN THE CENTRAL AUDITORY PATHWAY UPON UNILATERAL INTRA-COCHLEAR ELECTRICAL STIMULATION

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Within the last years cochlear implants have been applied successfully for the treatment of unilateral hearing loss, often with surprisingly high benefit. One reason for the successful treatment, including the relief of tinnitus, could be the normalization of spontaneous activity in the central auditory pathway as a result of the electrical stimulation.

The present study therefore, aimed to investigate, at a cellular level, the effect of a unilateral chronic intra-cochlear stimulation on key structures of the central auditory pathway. Normal-hearing guinea pigs were mechanically single-sided deafened. A standard HiFocus1j electrode array (on a HiRes 90k® cochlear implant) was inserted into the first turn of the cochlea. Four to five electrode contacts were then available for stimulation.

Six weeks after surgery, the speech processor (Auria®) was fitted based on tNRI-values and mounted on the animal's back.

Two experimental groups were stimulated for 16 h/day for 90 days, with HiRes®-strategy based on different stimulation rates: low rate, 275 pulses/second (p/s), high rate 5000 p/s. The results were compared to those of unilateral deafened controls, implanted but not stimulated. All animals experienced a standardised free field auditory environment.

The low-rate group showed a significantly lower average spontaneous activity bilaterally in the dorsal cochlear nucleus and the medial geniculate body than the controls. However, there was no difference in the inferior colliculus and the primary auditory cortex. Spontaneous activity of the high-rate group was also reduced bilaterally in the dorsal cochlear nucleus and in the primary auditory cortex. No differences could be observed between the high rate group and the controls in the contra-lateral inferior colliculus and medial geniculate body.

The present findings indicate a neuronal plasticity occurring to accommodate simultaneous electrical and acoustical stimulation. The unilateral intra-cochlear electrical stimulation seems to facilitate the homeostasis of network activity, since it decreases the spontaneous activity which is usually elevated upon deafferentiation. The electrical stimulation per se appears to be responsible for the bilateral changes described above, rather than the particular nature of the electrical stimulation (e.g. rate).

The results of the present study are of particular importance in cochlear implant patients with single-sided deafness. Based on the present findings an early implantation is suggested.

Advanced Bionics GmbH. (Sonova AG) supported this study.

M17: THE CONTRIBUTION TO BIMODAL BENEFIT OF DIFFERENT LOW FREQUENCY REGIONS IN CONTRALATERAL RESIDUAL HEARING

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The combination of acoustic and electric stimulation, called bimodal stimulation, can lead to large improvements when listening to speech in noise. Different hypotheses have been formulated to explain the cause of the bimodal benefit. For example, the acoustic signal may provide f0 information, facilitate glimpsing, or convey phonetic cues. These different possible sources of bimodal benefit will likely depend on different frequency ranges within the acoustic signal.

The present experiment investigates the individual contribution to bimodal benefit of different low frequency bands (B1: 20-280 and B2: 280-540 Hz). Speech reception thresholds (SRTs) were measured for IEEE sentences in multi-talker babble. Participants were normal hearing (NH) and cochlear implant (CI) listeners presented with real or simulated cochlear implant stimulation in one ear and acoustic stimulation in the other ear. In additional conditions, speech signals were processed through a vocoder preserving f0 and envelope information while discarding phonetic information. For NH listeners, higher levels of benefit were observed for the second band (280-540 Hz) than for the first (20-280 Hz). Discarding phonetic information severely decreases or abolishes the bimodal benefit. These results suggest that phonetic cues are more important to the bimodal benefit than the presence of the very low frequencies including f0. A large proportion of the CI participants tested did not show a benefit when one of the low frequency band was added to the electric stimulation. The results of CI listeners showing a benefit are compared to those of the NH listeners to determine whether they have similar trends.

Supported by Action on Hearing Loss

M18: COCHLEAR IMPLANTATION IN SINGLE SIDED DEAFNESS: BENEFITS IN SPEECH PERCEPTION AND SOUND SOURCE LOCALIZATION

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Introduction:Acquired single sided deafness (SSD) is not a rare condition. Until a few years ago, in these cases cochlear implantation was not considered a viable option due to the assumed much better hearing performance of the contralateral normal hearing ear. However, more recent observations with single sided deafness and modern implant systems were very encouraging. To assess the benefit of the CI, we conducted both cross-sectional and longitudinal studies.

So far, 29 patients with acquired single sided deafness were implanted. The subjects did not suffer from tinnitus, so their motivation to receive a CI and use the device was the wish to improve their hearing status, not to suppress a tinnitus.

Methods:12 of the subjects were tested in a cross-sectional study, 17 were followed longitudinally starting at the time of the first activation. Additionally, 4 recipients of a BAHA were tested using the same procedure. We also tested a reference group of 48 normal hearing subjects, subdivided into 6 age groups (10 year intervals). All subjects were native German speakers.

To assess potential benefits in speech perception, we employed a spatial release from masking paradigm, using a staircase procedure with OLSA nonsense sentences in two talker babble noise, where the SNR adaptively changed to determine the SRT. This procedure was performed in a free field setup with noise incidence angles 0°, ±45° and ±90°, in both (CI resp. BAHA) aided (CI+NH) and non-aided (NH) listening conditions, amounting to 10 individual test runs with SRTs as results. The SRT we found in the 90° (aided side) incidence angle condition was used to set two fixed SNRs and assess everyday HSM sentence and Freiburg monosyllabic words percentage scores in noise. In a subset of patients, we also tested speech and noise incidence angles reverted.

To assess sound localization abilities, we used a setup with nine equally spaced loudspeakers in the frontal horizontal plane (covering the ±90° angle range). A 1s CCITT noise burst served as the localization stimulus.

Results:Benefit of the CI+NH over the NH condition was greatest in the 45° and 90° (NH side) conditions. This benefit varied among subjects, but accounted to an average of a -5dB SRT shift. When presenting noise from the CI side angles, the benefit was smaller, however still could be demonstrated for some of the subjects. In the noise from front condition, we did not see a significant benefit.

Localization with the NH ear alone was negligible to poor, while in the CI+NH condition we observed fair to good performance. Localization benefit is statistically significant very shortly after device activation. Speech in noise benefit develops much more slowly over time, so that it becomes significant only after 12 months of CI use. In the cross-sectional study group, the speech performance benefit is statistically significant.

M19: AUDITORY EVOKED RESPONSES TO PITCH MATCHED STIMULI IN UNILATERAL COCHLEAR IMPLANT USERS WITH RESIDUAL HEARING IN THE CONTRALATERAL EAR

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It is now common for some cochlear implant (CI) users to have enough residual hearing in the unimplanted ear to compare the pitch percepts elicited by electrical stimulation with those elicited by acoustic hearing. We propose to use the changes in pitch percepts elicited by a given electrode as a metric to examine the perceptual process in adapting to the frequency mismatch between acoustic stimulation and the characteristic frequency of neurons stimulated by the electrode in quest, after implantation.

Here, we report on behavioral electroacoustic pitch matching data as well as on a potential physiological correlate of the behavioral data. Experiments were conducted using a real-time pitch matching platform that can be used with either Cochlear or Advanced Bionics (AB) patients. One part of the system uses the NIC V2 toolbox provided by Cochlear Americas to present interleaving short intervals (500ms) of acoustic and electrical stimulation to CI patients. Similar functionality is achieved for AB CI patients using a Harmony speech processor and a direct connect audio cable provided by Advanced Bionics Corporation. In both cases, CI patients are instructed to adjust the frequency of the acoustic tone to match the percept elicited by electrical stimulation. This procedure is repeated six times for several electrodes in the array. We have tested eight Cochlear CI patients and three AB CI patients. Nine of the eleven subjects have completed all trials for all active electrodes in their implants. All subjects tend to show tonotopic pitch percepts, but this does not happen for every electrode.. We also used the same stimuli while recording Auditory Evoked Potentials (AEP) from 10 CI patients using a Neuroscan system. The Cochlear CI patients received electrical stimulation in electrode 20 which is associated with a frequency band centered at 500 Hz in the clinical map and the other ear was stimulated with six different acoustic frequencies, ranging from 250 to 1000 Hz and including the frequency that was pitch matched to electrode 20 by each individual. The ABS CI patients were stimulated in similar manner with electrical stimulation in electrode 3 which is associated with a frequency band centered at 540 Hz in the clinical map. Each pair of electric and acoustic stimuli was repeated 500 times and the recording trigger was inserted at the onset of the acoustic stimulus. We also tested 10 normal hearing (NH) control subjects, where a fixed 500 Hz tone was presented to one ear (instead of the electrical stimulation) and tones of 250Hz, 375Hz, 500Hz, 625Hz and 1000Hz were presented to the other ear.

In the case of CI users, N1 latency was minimized when the acoustic and the electrical stimulus were pitch-matched. In the case of NH listeners, N1 latency was minimized when both ears were stimulated with the same frequency. These results suggest that the latency of N1 can potentially be used as an objective measure of pitch matching across two ears for the CI patients. We have started using this measure to examine adaptation to frequency mismatch over time in two CI patients.

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M20: RATE OR PLACE: WHICH IS THE REAL PITCH?

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In the cochlea of a normal hearing listener, frequency information is coded tonotopically such that lower frequencies are presented to the cochlear apex while higher frequencies are presented towards the base. Each frequency has its own unique location in the cochlea. Additionally, each location in the cochlea is vibrated at the frequency corresponding to that location. As a result, pitch could be coded by the location of the cochlea stimulated (place), the frequency at which the cochlea is vibrated (rate), or a combination of the two. With normal hearing listeners, it is very difficult to independently manipulate rate and place information to determine the relative importance of each variable in coding pitch.

With cochlear implants, rate and place cues are easily independently manipulated. Any electrode can be stimulated at any rate. On a single electrode, increasing the rate of stimulation is reliably reported as producing a higher pitch. Similarly, switching from one electrode to another located more basally in the cochlea is also reliably reported as producing a higher pitch. When an implant user is presented with conflicting changes in rate and place, pitch ranking is made more difficult. Therefore, it is clear that both rate and place contribute to a pitch percept.

Using Multi-Dimensional Scaling, it has previously been shown that a change in rate and a change in place represent two independent perceptual dimensions. Therefore, although both rate and place represent perceptual dimensions labeled as pitch, they do not represent the same pitch quality. Unfortunately, with cochlear implant users, it is very difficult to determine which of these two types of pitches (rate or place) are more like pitch as experienced by normal listeners.

A new patient population with normal hearing in one ear and a long (31 mm) Med-El cochlear implant array in the contralateral ear provide an excellent opportunity to compare perceptual changes from rate and place with electrical stimulation with the perceptual change elicited from a change in acoustic frequency. Using a multi-dimensional task, we compared the perceptual relationships between a change in rate, a change in place, and a change in acoustic frequency. Acoustic stimuli consisted of pure tones at 75, 150, and 300 Hz. Electric stimuli consist of three adjacent electrodes with amplitude modulations at either 75, 150, or 300 Hz. The experiment was repeated in both the apex and the middle of the array. Results consistently indicate that in the middle of the array, a change in acoustic frequency is perceived as a change in place of stimulation. Results in the apex are less consistent, but suggest that most subjects perceive a change in acoustic frequency as a change in rate. The data suggests that while place and rate pitch are different, the dimension that represents a change in acoustic frequency varies across the array. Furthermore, it suggests that to provide a natural pitch sensation with a cochlear implant, it is important to provide temporal pitch in the apex, but not across the entire cochlea.

M21: CONFLICTING CUE VOWEL IDENTIFICATION BY BIMODAL CI USERS AND BY NORMAL HEARING LISTENERS USING AN ACOUSTIC MODEL OF A CI AND CONTRALATERAL HEARING AID

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For many bimodal cochlear implant (CI) users, the combined input of a CI and contralateral hearing aid (HA) provides significant bimodal benefit, i.e. improvement in speech scores relative to when either of these devices is used alone. However, bimodal benefit is not observed in all bimodal CI users, and in some cases, the combined input of a CI and contralateral HA results in degradation of speech scores, suggesting that the two devices can interfere with each other. To investigate bimodal integration of speech cues, or lack thereof, in bimodal CI users, an identification paradigm was employed using non-conflicting and conflicting cue combinations of the synthetic vowels /i/, /a/, and /u/. The non-conflicting stimuli comprised presenting the same vowel to each ear, and the conflicting cue stimuli comprised presenting one vowel to the CI, and a different vowel to the HA. Subjects were asked to categorize what they heard using nine possible responses representing the most common monophthongal English vowels. Subjects' responses to the conflicting cue vowels varied. Some listeners' responses primarily followed the electric input and other listeners' responses primarily followed the acoustic input. There were other subjects whose response patterns were more complex, e.g., consistently responding /u/ when the acoustic stimulus was /i/ and the electrical stimulus was /a/, akin to the McGurk effect.

To further investigate the above results, a similar paradigm was employed with normal hearing listeners using an acoustic model of a CI and contralateral HA. The synthetic vowels /i/, /a/, and /u/ were processed using a 6-channel noise-band vocoder for the CI simulation, and using HeLPS software by Sensimetrics for the HA simulation (moderate-to-severe sloping loss). Non-conflicting and conflicting cue combinations of processed vowels, one from the CI simulation and one from the HA simulation, were presented in dichotic and diotic conditions. Subjects were required to classify what they heard using the nine possible responses offered to CI subjects described above. Subjects were given the option of choosing one response if they heard one, fused vowel, or two responses if they heard two separate vowels. Under dichotic conditions, the large majority of subjects consistently heard two distinct vowels, even when the same vowel was presented to each ear. There was a consistent trend where responses to one vowel (whether processed through the CI or HA simulation) appeared to be influenced by the vowel presented to the contralateral ear. However, the particular vowel for which this occurred varied across subjects. Under diotic conditions, many subjects reported hearing two distinct vowel sounds, but there were a number of responses to conflicting cue stimuli that were perceived by many subjects as one fused sound, suggesting bimodal integration. These results are informative in helping us understand how bimodal CI users integrate speech information between their CI and HA devices.

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M22: EFFECTS OF CHRONIC ELECTRO-ACOUSTIC STIMULATION ON CORTICAL RESPONSES

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Cochlear implant (CI) recipients with residual hearing in their implanted ear have access to information from both acoustic hearing (via their remaining low-frequency hearing) and electric hearing (via their CI stimulating the more basal, high frequency region of the cochlea). The aim of this project was to investigate, in an animal model, the effects of long-term partial deafness and chronic electro-acoustic stimulation on the representation and integration of stimuli from the two modalities in primary auditory cortex (AI) of the cat.

Neonatal cats (n=14) were deafened with a singlesubcutaneous administration of kanamycin sulfate (KA; 300 mg/kg) and intravenous infusion of ethacrynic acid (25 mg/kg) 5days after birth. At two months of age, 10 neonatally deafened animals were implanted with a Hybrid-L 14 electrode array. Adult cats (n=3) were deafened withdaily subcutaneous injections of KA (200 mg/kg) for upto 30 days. Six months after the deafening, all adult deafened animalswere implanted with a Hybrid-L 14 electrode array.One month after implantation, chronic intracochlear electricalstimulation from a clinical CI and speechprocessor commenced and continued for 5 months.At the end of the chronic stimulation period, we recorded from single- and multi-unit clusters (n =1034) in AI using multi-channel silicon electrode arrays.

The proportion of units responsive to electric only (E), acoustic only (A) or both acoustic and electric (EA) stimuli was significantly (Chi-squared, $p < 0.05$) affected by the animals hearing history (neonatally deafened unstimulated: E= 7%, A= 19%, EA= 74%; neonatally deafened chronically stimulated: E= 13%, A= 8%, EA= 79%; adult deafened chronically stimulated: E= 9%, A= 27%, EA= 64%). These results indicate that chronic CI use in neonatally deafened animals can restrict the extent of AI that can be activated by acoustic stimuli. Responses to combined electro-acoustic stimulation were also recorded and the complex interactions of the two stimuli at individual recording sites will be discussed.

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M23: THE ROLE OF PITCH CUES FROM CONTRALATERAL RESIDUAL ACOUSTIC HEARING IN COCHLEAR IMPLANT USERS' PERCEPTION OF INTONATION

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In some cochlear implant (CI) users contralateral residual low-frequency acoustic hearing provides benefits to speech perception. There are several possible sources of this bimodal benefit, including additional low-frequency spectral resolution, enhanced glimpsing in noisy conditions, and better perception of voice pitch. Here the contribution to intonation perception of voice pitch cues from residual acoustic hearing was assessed.

Tasks were the identification of sentences as either question or statement and identification of the stressed word within sentences. Each task was carried out in quiet and in noise, with naturally spoken stimuli and with stimuli processed to ensure that only F0 cues were available. Stimuli were presented free field to listeners using their everyday CI and hearing aid settings. In addition, the ability to detect pitch changes through the CI and through residual hearing was assessed from detection thresholds for 2-Hz sinusoidal frequency modulation (FM) applied to the F0 of vowel-like harmonic complexes.

Stressed word identification in quiet was at ceiling with CI alone for naturally spoken stimuli, but well below ceiling for processed F0-alone stimuli, suggesting a major role for non-pitch cues (e.g., duration) in conveying stress. Question/statement identification was generally below ceiling in quiet and similar for both natural and F0-alone stimuli. In both intonation tasks in quiet, performance over the group was significantly better in bimodal conditions than with CI alone. However, the amount of bimodal benefit varied substantially across individuals, with several listeners showing no benefit. There was little evidence of bimodal benefit in noise. There was not a close relationship between acoustic FM detection thresholds and the degree of bimodal benefit for intonation perception. While those subjects with particularly poor FM detection did not show bimodal benefit, relatively good FM detection could occur alongside an absence of bimodal benefit. This suggests that good detection of pitch changes via acoustic hearing is necessary but not sufficient for bimodal benefit in intonation perception. For some bimodal CI users with relatively good acoustic pitch perception there may be difficulty in integrating voice pitch information across modalities.

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M24: COCHLEAR IMPLANTATION IN ADULTS WITH ACQUIRED SINGLE-SIDED DEAFNESS: A PROSPECTIVE UK TRIAL

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Binaural hearing underpins the ability to localise sounds and is beneficial when listening to speech in the presence of other spatially-separate sounds such as competing talkers or noise. These spatial listening abilities are severely compromised or absent in individuals with a single-sided deafness (SSD) for whom useful acoustic information is provided by one ear only. Consequently, individuals with SSD report difficulties with listening and communicating in many everyday listening situations and report an associated decrease in quality of life.

The current standard of care for those individuals who experience listening difficulties as a result of SSD is a Contra-lateral Routing of Signals (CROS) hearing aid. A CROS aid improves the audibility of signals on the impaired side of the head by diverting acoustic information arriving at the impaired ear to the non-impaired hearing ear. Thus, while a CROS aid improves access to information on the impaired side, it does not convey the binaural cues which underpin spatial listening abilities. Recent evidence suggests that cochlear implantation may be an effective intervention for SSD as it has the potential to restore access to binaural cues such as inter-aural level differences and support useful aspects of binaural hearing.

The present study is a longitudinal within-subjects comparison of the benefits of CROS aid use and the benefits of cochlear implantation in adults with an acquired single-sided deafness. Benefit will be assessed in terms of laboratory tests of binaural hearing and self-report measures of listening difficulty and health-related quality of life. Ten adult participants will be recruited across five UK cochlear implant programs. A two-month baseline assessment will be followed by a three-month trial of a CROS aid, after which participants may elect to receive a cochlear implant. Behavioral measures of localisation in anechoic and reverberant conditions and of speech perception in noise will be obtained at baseline, at the end of the CROS trial, and nine months after cochlear implantation. Self-report measures of listening difficulty and health-related quality of life will be obtained twice at baseline, after one and three months of CROS use, and three and nine months after cochlear implantation.

This work is supported by infrastructure funding from the National Institute for Health Research and by Cochlear Europe Ltd.

M25: SPEECH RECOGNITION IN NOISE AND LOCALIZATION ABILITIES IN ADULTS WITH UNILATERAL HEARING LOSS: IMPLICATIONS FOR COCHLEAR IMPLANT CANDIDACY

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Despite coping strategies, deficits of unilateral hearing loss (UHL) are not easily overcome. Sound localization and listening in noisy environments are especially problematic. Accompanying the numerous negative consequences of UHL is considerable unexplained individual variability in the magnitude of its effects. Identification of co-variables that affect outcome and contribute to variability in UHLs could augment counseling, treatment options, and rehabilitation. Cochlear implantation as a treatment for UHL is on the rise yet little is known about factors that could impact performance or whether there is a group at risk for poor cochlear implant (CI) outcomes when hearing is near-normal in one ear. The overall goal of our research is to investigate the range and source of variability in speech recognition in noise and localization among individuals with severe to profound UHL and thereby help determine factors relevant to decisions regarding cochlear implantation in this population.

The present study evaluated adults with severe to profound UHL and adults with bilateral normal hearing (NH). Measures included localization, roving-source speech recognition (words from 1 of 15 speakers in a 140° arc) and adaptive sentence understanding (R-Space and restaurant noise). There were three age-gender-matched groups: NH listening bilaterally (NHBil), NH listening with one ear with the opposite ear blocked (NHUni), and UHL listening with the intact ear. NHBil were significantly better than UHLs on all measures. Performance variability was greater in UHLs than NHBil participants. NHUni had better monosyllabic word understanding than UHLs for words presented on the blocked/deaf side but not the hearing side. In contrast, UHLs localized better than NHUni for stimuli on the hearing side but not the blocked/deaf side. This suggests UHLs had learned strategies for improved localization on the side of the intact ear. In addition, UHL participants with childhood HL onset localized significantly better than those with recent HL onset however, these two groups did not differ for speech recognition in noise. Age at onset in UHL adults appears to affect localization ability differently than speech in noise. These study findings may help differentiate the effects of age at onset and length of HL for potential UHL CI candidates from more traditional CI candidates. In addition, results support the need for a broader range of measures for use in CI candidacy evaluations as individuals with unilateral severe to profound HL and varying amounts of hearing in the “good ear” are being considered potential candidates.

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M26: CORTICAL BINAURAL INTERACTIONS AND ITD REPRESENTATION OF CATS WITH CONGENITAL SINGLE-SIDED DEAFNESS

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Congenitally deafness leads to various functional deficits in the auditory cortex such as decrease in binaural sensitivity (Tillein et al., 2010, Cereb. Cortex, 20:492). The present study investigated the effects of early single-sided deafness (SSD) on cortical binaural sensitivity. Two adult cats with congenitally unilateral deafness were compared to five adult deaf and hearing controls. Controls were acutely deafened by intracochlear application of neomycin. All animals were acutely stimulated with charge-balanced biphasic pulses (200µs/phase) in wide bipolar configuration through a custom made cochlear implant inserted into the scala tympani on either side. Cortical regions which were most responsive (defined by local field potential mapping) were subjected to recordings of multi-unit activity using 16 channel multielectrode arrays.

Binaural interactions of EE type were classified as facilitation (binaural response > 120% of the sum of monaural responses) and occlusion (binaural response < 80% of the monaural responses). Sensitivity to ITDs between -600 and +600 µs was tested with pulse trains (500 Hz, 3 pulses) at levels of 0 - 10 dB above brainstem response thresholds. Spike rates were evaluated as a function of intensity and ITDs.

Within the SSD group, responses were dependent on the recorded hemisphere (relative to the hearing ear). Ipsilateral to the hearing ear, a massive increase of ipsilateral excitation was found. Threshold saturation and dynamic range were similar to hearing controls. A significant increase in responsive sites was observed in SSD group, so that with respect to responsiveness single-sided animals compared well to normal hearing controls. In consequence, cortical cells remained responsive to binaural stimulation, although the response was not ITD specific. Modulation depth of ITD functions in classified responses was additionally significantly smaller than in deaf and in hearing animals.

In consequence, monaural deprivation leads to a distinct decrease of sensitivity to variation of binaural cues, whereas general responsiveness to stimulation is comparable to hearing animals. This finding is in strong contrast to the situation in binaural deafness and demonstrates the detrimental influence of unilateral deafness on binaural feature sensitivity and consequently on the localization ability.

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M27: ELIMINATION OF OUTER HAIR-CELL GAIN EXPLAINS THRESHOLD CHANGES IN COCHLEAR IMPLANTED EARS

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We reviewed hearing threshold data collected during recent clinical studies of residual hearing patients who were implanted via round window with CI422 Slim Straight (N=33) or Nucleus Hybrid L24 arrays (N=66). The Slim Straight array was inserted linearly 20 to 25 mm to obtain insertion angles from ~360° to ~450°, measured individually from CT scans, and the Hybrid array was inserted to 16 mm to obtain ~270°.

The published data for these “straight” arrays indicates that subjects with low thresholds tend to lose more hearing after implantation than those with greater hearing loss (HL) (Skarzynski et al, 2012, Audiol Neurootol, 17, 82-91; Lenarz et al. 2009, Audiol Neurootol, 14 Suppl 1, 22-31). We hypothesized that the presence of the electrode array may interfere with the “active” function of outer hair cells (OHCs) which is essential in providing low auditory thresholds.

Histological and imaging studies show that these straight arrays take up a position along the lateral wall very close to, or even in contact with, the organ of corti. Thus we may expect some mechanical interference with the movement of the basilar membrane (BM) which is assisted by motile activity of OHCs for low inputs. The gain at threshold provided by OHCs may be derived from input/output functions estimated either from BM velocity in animals (e.g. Ruggero et al, 1997, J Acous tSoc Am, 101, 2151-2163) or from psychophysical forward masking in humans (Moore, 2012, Introduction to the Psychology of Hearing, 67-133). In normal hearing subjects maximum gain is estimated to be ~60 dB and this reduces in a nearly linear fashion to zero as HL increases to ~90 dB, with slope approximately -½ dB gain per 1 dB increase in HL.

For the Slim Straight array where the tip of the electrode was located at, or apical to the cochlear position corresponding to the test frequency, based on the Greenwood CF function, threshold increases were significantly, negatively correlated with pre-operative thresholds ($r=0.7$, $p<0.0001$). The slope of the linear regression line was -0.48, with intercept 55 dB, in quantitative agreement with the OHC gain function. Where the electrode tip was >135° basal to the test frequency, threshold increase was not significantly influenced by pre-operative threshold ($r=0.05$, $p=0.69$). Thus for frequencies ½ turn apical to the electrode tip there was no apparent elimination of OHC function, with a mean threshold increase of 10 dB (NS >0.0). Results for Hybrid array were in agreement with the Slim Straight with significant correlations for tip-to-CF ~0° ($r=0.4$, $p=0.015$) and no significant correlation for tip-to-CF ~190° ($r=0.09$, $p=0.46$).

Clinical trials were sponsored by Cochlear Europe.

M28: USING TACTILE AIDS TO PROVIDE LOW FREQUENCY INFORMATION FOR COCHLEAR IMPLANT USERS

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This research asks whether tactile aids, by providing low-frequency information, could benefit unilateral cochlear implant (CI) users.

A large literature documents that low-frequency acoustic information from the ear contralateral to a CI can provide, in the best cases, up to 40 percentage points improvement in speech understanding score. However, only a little over half of CI patients have residual hearing in the ear contralateral to the implant. An even small number have hearing in the implanted ear.

Traditionally tactile aids were used as an aid to lip-reading and have been found useful for localization, environmental sound identification, syllable and stress pattern discrimination and word learning. Tactile sensation has a frequency range approximating that of patients with only low-frequency hearing, i.e., at 125 and 250 Hz. Just as patients with only low-frequency acoustic hearing benefit from acoustic stimulation, CI patients may benefit from tactile stimulation.

The test materials were sentences developed by J. Liss (see Spitzer et al., 2009) to probe the value of acoustic cues, such as pitch prominence, amplitude envelope and vowel length, for the detection of lexical boundaries. Unilateral CI users and bimodal CI users who were temporarily deprived of the benefit of a hearing aid participated in the present study. Wide-band acoustic stimuli were presented to CI side through a loudspeaker at subject's most comfortable level. The subjects held a bone-anchored hearing aid (BAHA) to receive simultaneous tactile stimuli -- a carrier signal modulated with the envelope of the acoustic signal. The test conditions compared the benefit provided by each of the combination of cues in the following conditions: 1) CI only, 2) CI + tactile aid, 3) CI + tactile aid + visual information and 4) CI + visual information. Preliminary results suggest that tactile input can be of benefit to unilateral CI patients.

Support provided by NIDCD Grant R01 DC 010821 to MFD.

M29: MULTI-ELECTRODE LATERALIZATION USING INTERAURAL LEVEL DIFFERENCES IN BILATERAL COCHLEAR IMPLANT SUBJECTS

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An improved ability to localize sound is one of the advantages of bilateral cochlear implantation. Conventional cochlear-implant (CI) mapping does not consider possible binaural consequences like the introduction of spurious interaural level differences (ILDs). Moreover, real world sounds activate multiple electrodes, which can introduce additional variability in the encoding of ILDs due to spread of excitation. We tested if inconsistent lateralization cues are introduced by the mapping process for both single- and multi-electrode stimulation and whether removing spurious ILDs (i.e., “centering” sound images) on single-electrode pairs improves lateralization for multi-electrode stimulation.

Six bilateral CI users were tested in a lateralization experiment using direct stimulation with monopolar, 1000 pps, 500 ms, constant-amplitude pulse trains. Thresholds and most-comfortable "C" levels were obtained for 5 electrodes in each ear using a conventional mapping procedure. Stimuli were presented bilaterally at 70-80% of the dynamic range for combinations of 1-5 electrode pairs and ILDs of -20 to +20 current units (CUs) were imposed. The subjects reported an intracranial image lateralization for the single- and multi-electrode stimulation.

Only 30% of the auditory images were perceived as centered (lateralization offset magnitude <2 CUs) for the single-electrode pairs. Similarly, only 30% of the auditory images were perceived as centered for the multi-electrode pairs. Therefore, presenting stimuli at a fraction of the "C" level in each ear often does not ensure the perception of a centered auditory image.

Multi-electrode combinations were retested after centering single-electrode pairs. A 20% or greater decrease in the lateralization offset magnitude was observed in 12 out of 29 multi-electrode combinations, although in three of them level adjustments caused the perception of the auditory image to move across the midline to the opposite side of the head. Only significant level adjustments, mostly on electrode pairs with an offset > 9 CUs (83%), produced systematic changes of the lateralization offset magnitudes in the multi-electrode stimulation. Therefore, future bilateral mapping strategies might consider focusing on reducing spurious ILDs by centering the single electrodes with the largest lateralization offset magnitudes. Such a mapping strategy may be a clinically-based tangible step in improving localization performance in bilateral CI users.

This study was supported by NIH-NIDCD R00-010206 (Goupell).

M30: RATIONALE AND IMPLEMENTATION OF A BIMODAL FITTING FORMULA

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Introduction
The effectiveness of cochlear implants has increased markedly over the past years. As a consequence, candidacy for implantation has been relaxed and more CI users have functionally useful contralateral hearing. Such bimodal users often report improved speech understanding, particularly in noisy situations, better tonal quality and improved lateralization of sound sources. The individual bimodal benefit, however, varies considerably. A potential reason for this is that conventional hearing aid fitting does not account for certain specific characteristics of bimodal listening. Of particular importance are: low-frequency audibility, spectral overlap of electric and acoustic stimulation, loudness balance, inter-aural tonotopy and (dynamic) synchronization of adaptive signal processing. While many of these requisites are highly individual and thus, may have to be addressed by individual fine-tuning, others might be accounted for, at least partially, by a prescriptive fitting formula specifically designed for bimodal listeners.

Method
In order to improve the efficiency of bimodal fitting, a bimodal fitting formula is being developed that aims at accounting for the specific characteristics of bimodal listening.
This formula can use any established fitting formula (e.g. DSL v5, NAL-RP) as a starting point. In the current implementation, Phonak's proprietary formula (Adaptive Phonak Digital) is used. The importance of low-frequency audibility is addressed by maximizing effective audibility [1] based on the individual audiogram. Edge frequencies of Dead Regions can be either entered manually or are estimated automatically from the audiogram. As a second step, the static and dynamic behavior of the CI's AGC has been approximated by the hearing aid's compression system to account for loudness balance and dynamic synchronization between electric and acoustic hearing. This is accomplished by a special parameterization of channel-coupling, time constants, compression knee-points and compression ratios in the hearing aid.

Results
Compared to conventional hearing fitting aid formulae, the novel bimodal fitting formula typically results in decreased gain at high frequencies and increased gain at low frequencies. Technical measurements using percentile analysis show that the dynamic behavior of the CI's AGC can be approximated with the HA's AGC for speech signals. First results from studies using the bimodal fitting formula indicate that this formula can improve speech understanding in noise and provides a good starting point for individual fine-tuning.

[1] TYC Ching, H Dillon, R Katsch, D Byrne, 2001. Maximizing Effective Audibility in Hearing Aid Fitting.EAR & HEARING, VOL. 22 NO. 3

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M31: SOME ACOUSTIC MODELS OF COCHLEAR IMPLANTS ARE QUITE GOOD BUT THE MOST COMMONLY USED ONES ARE NOT: A SINGLE-SIDED DEAFNESS STUDY

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Acoustic models are commonly used to simulate the percepts elicited by cochlear implants (CIs). Front-end processing in these models is very similar to that used in CIs, and the percepts that would be caused by electrical stimulation in a real CI are simulated by either noise bands or sinusoids. In “standard” acoustic models the noise bands or sinusoids have center frequencies identical to those of the analysis filters. Acoustic models are very useful for experiments that cannot be done or would be impractical with actual CI users, and can also be used as informative control conditions in CI experiments, Despite their usefulness such models have never been convincingly validated and the few studies that addressed this issue suggest that they are imperfect at best. However, a new opportunity to validate acoustic models of CIs by doing a direct perceptual comparison is provided by “Single-sided deafness” (SSD) patients, those who have a cochlear implant in one ear and have near perfect hearing in the other ear.

Three different methods were used. First, four cochlear implant users with SSD were presented the same sentence twice: the unprocessed sentence was sent via a direct connection to their CI speech processor and after a short pause the sentence was processed through a cochlear implant acoustic model and presented to the normal hearing ear through loudspeakers. The analysis filters used in the acoustic model had the same frequency boundaries as those in the listener’s speech processor, but the noise bands of the vocoder were adjustable. Listeners used a graphical interface to determine a “self-selected model” by modifying the lowest and highest frequencies spanned by the noise bands until the noise vocoder sounded as similar as possible to the percept provided by the CI. The process was then repeated for a sinewave vocoder. Then, listeners answered questionnaires about the similarity between the sentences as heard through the cochlear implant and through four different types of models: noise vs. sinewave, and standard vs. self selected. The third method consisted of comparing speech perception scores obtained with the CI and with the acoustic models.

Self-selected models produced sounds that were rated “very similar” to those heard through the implant, but for three out of four subjects these self-selected models were quite different from the standard models. The latter were rated less similar to the CI and resulted in higher speech perception levels than the CI. This suggests that the widely used standard acoustic models are inaccurate representations of the percepts actually heard by some cochlear implant users.

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M32: EFFECTS OF COCHLEAR IMPLANT LOWER CUTOFF FREQUENCY ON SPEECH PERCEPTION IN BIMODAL COCHLEAR IMPLANT USERS

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The recent implantation of individuals with more residual hearing has led to the use of "acoustic+electric" (A+E) stimulation, where a cochlear implant (CI) is worn with a hearing aid (HA) in the non-implanted ear (bimodal stimulation), or even in the same ear (Hybrid stimulation). A+E stimulation can improve speech perception in quiet and in noise, as well as sound localization (review: Ching et al., 2007). However, the amount of benefit is variable across individuals. The purpose of this study was to determine whether raising the lower frequency cutoff encoded by the CI improves speech perception benefit in noise for bimodal CI users, as recent studies in Hybrid CI patients suggest that reducing the overlap between the electric and acoustic frequencies may improve the speech perception benefits in noise (Reiss et al,, 2012; Vermiere et al., 2008).

Nineteen experienced CI users were recruited for speech perception tests. Eleven subjects were tested in the bimodal condition and ten subjects were tested in the CI-only condition. Speech processors were programmed with four experimental programs in which the low frequency cutoffs were: 188, 438, 688, and 938 Hz. To minimize the need to adapt to spectral shifts, frequency-to-electrode allocations were maintained by deactivating electrodes. In the bimodal condition, subjects were tested with their own HAs when possible to minimize the need to adapt to a new HAs; however, if HAs did not meet NAL-NL1 targets, they were programmed with a lab hearing aid. The effects of cutoff frequency on speech perception were evaluated based on scores on consonant-nucleus-consonant (CNC) and AZBio sentences in background babble, and signal-to-noise ratios (SNRs) for spondee words in background babble. Performance in the bimodal condition was compared to the CI-only condition to determine whether effects were due to the reduced low frequency information provided to the CI, or to interactions of the CI with the acoustic hearing.

Subjects tested in the CI-only condition showed significant group decreases in speech recognition performance as the cutoff frequency increased. In contrast, subjects tested in the bimodal condition showed more variable patterns as the cutoff frequency increased. Half of the subjects in the bimodal condition performed best with an intermediate cutoff, i.e. less overlap between acoustic and electric frequencies, while the other half performed best with the lowest cutoff, i.e. full overlap. This variability may reflect differences in residual hearing and HA type. The results overall suggest that for some subjects, less overlap is better for speech perception in quiet and in noise.

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M33: MULTISENSORY INTEGRATION IN ADULT COCHLEAR IMPLANT USERS WITH HEARING PRESERVATION

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Research has shown that hearing preservation in the implanted ear yields significant benefit for speech recognition in complex environments (e.g., [Dunn, Perreau, Gantz, and Tyler, 2010, J. Am. Acad. Audiol., 21(1), 44-51.][Gifford, Dorman, Skarzynski, Lorens, Polak, Driscoll, Roland, and Buchman, 2013, Ear Hear, Epub ahead of print 26 Feb 2013.]). While significant, the effect size is relatively small, ranging from 10- to 12-percentage points. Patient reports, however, are overwhelmingly positive regarding improved ease of listening and communication.

Communication is a multisensory experience, involving integration of auditory of visual stimuli. A Bayesian model of multisensory integration suggests that we weight sensory cues based on reliability of the signal (Deneve, Pouget, 2004, J. of Physiology-Paris, 98(1), 249-258). Hearing preservation in the implanted ear may increase the perceived reliability of the auditory signal, potentially increasing audiovisual integration. Thus the purpose of this study was to evaluate whether preserved hearing in the implanted ear yielded greater audiovisual integration than the bimodal condition.

We utilized a within-subjects, repeated-measures design, to assess audiovisual integration in nine adult participants. Tasks included the McGurk effect and simultaneity judgments (SJ) for audiovisual stimuli (onset asynchronies +/-500 ms). For the McGurk effect, participants exhibited greater audiovisual integration in the best-aided condition (20% increase), consistent with the Bayesian model of multisensory integration and providing empirical support of patients' subjective reports. For the SJ task, however, the width of the temporal binding window (TBW) "typically inversely proportional to the strength of integration"was consistently wider for the best-aided condition. Binaural summation associated with the best-aided condition may have affected perceived loudness for the low-frequency tone. Greater loudness perception influences perceived tonal duration, having the potential to confound the SJ results. Future research is warranted to explore this unexpected result.

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M34: INVESTIGATION OF THE TOP-DOWN PROCESS IN SIMULATED COMBINED ELECTRIC AND ACOUSTIC STIMULATION

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Speech communication requires listeners to encode sensory information and to use higher-level cognitive processes to decode the extracted information. Previous studies have shown that the benefit of combined electric and acoustic stimulation (EAS) is typically greater at the sentence level than at the phoneme or word level. A possible explanation for this difference is that linguistic context contributes to EAS benefit, and sentence stimuli provide greater amounts of context than phonemes or words. For example, using a phoneme restoration paradigm, Baskent (2012, JARO, 683-692) reported a slightly greater top-down repair effect on sentence recognition in simulated EAS compared to simulated cochlear implant stimulation (vocoder alone). The purpose of the present study was to quantify the extent to which low-frequency speech or periodicity cues enhance listeners' use of top-down linguistic knowledge in simulated EAS at the phoneme, word, and sentence levels.

Two experiments were conducted on young normal-hearing listeners. Each subject listened to stimuli in three conditions: vocoded speech alone (V), low pass speech alone (LP), and V + LP. In the first experiment, speech materials included nonsense syllables and CVC words that were identical to those used in Boothroyd and Nittrouer (1988, JASA, 101-114), as well as CUNY sentences. The low-pass cutoff was fixed at 500 Hz with attenuation rate of 60 dB per octave. The number of noise-vocoded channels was varied to obtain different levels of performance. The context effect was quantified using the mathematical model proposed by Boothroyd and Nittrouer (1988). In the second experiment, high-context (CUNY) and low-context (IEEE) sentences spoken by the same female speaker were presented to each subject in quiet for the V and V + LP conditions. Speaking rate was held constant for both sets of sentences. The original speech was subjected to 2- and 4-channel noise-vocoder processing. A harmonic complex was created to preserve the fundamental frequency contour of the original speech. The harmonic complex was then low-pass filtered at 500 Hz and modulated with the temporal envelope of the low-pass speech.

Findings indicate a similar context effect at the phoneme and word levels for the V and V + LP conditions. However, at the sentence level, the simulated EAS benefit was significantly greater for CUNY sentences compared to IEEE sentences. This pattern of results was observed when comparisons were made at approximately equal baseline (V condition) performance levels (20-30% correct) (i.e., 4ch-vocoder IEEE vs. 2ch-vocoder CUNY), or for an equal number of channels (i.e., 2ch-vocoder for both IEEE and CUNY). These findings suggest that the quality of the sensory input affects top-down processing for speech recognition, and that EAS benefit is further enhanced in everyday communication, which inherently includes linguistic context.

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M35: DEVELOPMENT OF CORTICAL PROCESSING OF CHANGES IN PIANO TONES IN EARLY-IMPLANTED CHILDREN; CONNECTIONS TO SINGING AND PERCEPTION OF SPEECH IN NOISE

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Introduction: Cochlear implant (CI) provides weak cues to F0 and timbre. We examine cortical responses in early-implanted children to investigate the development of detection of F0 and timbre changes as well as the hypothesis that singing could enhance perception of F0, timbre and speech in noise.

Methods: Unilateral CI users and normal hearing (NH) controls were aged 4-14. Changes in F0, musical instrument, duration, intensity and temporal gaps were presented in a rapid multi-feature paradigm amongst 295 Hz natural piano tones. Also the ability to detect sentences in background noise was investigated. Measurements were conducted twice. 12 from total of 21 CI children sang weekly at home (7 of these at least 5 times per week). They did not differ from other CI children in the exposure to other musical activities.

Results: For the CI group mismatch negativity (MMN) to deviants from piano standards showed poorer detection of changes in instrument (timbre), duration and the presence of gaps than NH group. P3a suggested less accurate attention switching to change of instruments (timbre) in CI group and different development of attention allocation between groups. Cortical responses for F0 deviants were surprisingly similar between CI and NH groups. Compared to the other early-implanted children, the CI children who sang regularly had earlier MMN and P3a for changes in F0, earlier MMN for change from piano to cembalo and their P3a developed more. They were also better in perception of speech in noise which was connected to cortical detection of changes in F0 and timbre.

Conclusions: Early-implanted children have difficulties or delay in processing of gross temporal changes in piano tones and deficits in perception timbre. Future studies should investigate possible causal links in CI children from singing and other musical activities to enhanced auditory perception and attention.

Torppa, R., Salo, E., Makkonen, T., Loimo, H., Pykäläinen, J., Lipsanen, J., Faulkner, A. &Huotilainen, M. (2012). Cortical processing of musical sounds in children with Cochlear Implants. Clinical Neurophysiology, 123, 1966-79.

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M36: VISUALIZATION OF HUMAN INNER EAR ANATOMY WITH HIGH RESOLUTION 7 TESLA MAGNETIC RESONANCE IMAGING

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Cochlear implantation requires detailed information of the microscopic anatomy for surgical planning, morphological calculations and to predict functional success.In many centersMRI of the inner ear and auditory pathway - performed on 1,5 or 3 Tesla systems- is part of the preoperative work-up of CI-candidates. A higher magnetic field strength results in a higher signal-to-noise ratio that can be used for more detailed imaging than previously possible. The visualization of delicate and small-sized inner ear structures might benefit from such higher resolution imaging. However, due to technical complexity, protocols need to be specifically developed for higher field scanners. The aim of this study was to investigate the feasibility of representing inner ear anatomy on images performed with 7 Tesla MRI.

A high resolution T2-weighted protocol of the inner ear was developed in healthy volunteers on a 7 Tesla MRI system (Philips Healthcare, The Netherlands), with an isotropic resolution of 0.3 mm, resulting in a scan duration of 10 minutes. Two high permittivity pads [1], which consisted of a deuterated suspension of barium titanate ($\varepsilon_r \approx 290$), were designed using numerical simulations and positioned next to the ears to enhance the signal at the location of the inner ear. The optimized protocol has been applied to CI candidates and was compared to the standard pre-operative 3 Tesla MRI work-up. Up to now five subjects have been included in this study.

The use of dielectric pads was essential in obtaining a high image quality. Without these pads, the magnetic field inhomogeneities result in signal loss in the area of interest. In CI candidates, the images obtained at 7 Tesla showed improved contrast and spatial resolution in comparison to the regular 3 Tesla protocol.

Fine intra cochlear anatomical structures, such as modiolus, interscalar septum and spiral osseous lamina were identified in high detail. This resulted in a clear delineation of scalavestibuli and scala tympani within the 2.5 cochlear turns. The isotropic resolution is allowed for oblique multiplanar reconstruction, without the loss of image quality and resulted in excellent visualization of the different branches of the cochleovestibular nerve. These high resolution images can be used for individual planning of CI and when registered with post-operative CT images will contribute to the evaluation of cochlear trauma in individual patients. This is subject of ongoing research at our center.

The high resolution achievable with 7 Tesla MRI enables and even substantially improves the representation of the inner ear anatomy and can contribute to improve preoperative planning of cochlear implantation.

This study was financially supported by Advanced Bionics and the Heinsius-Houbolt Fund

References: [1] Teeuwisse et al., MRM 2012, 67:1285–1293.

M37: SOURCE ANALYSIS OF EASSR GENERATORS MODULATED AT SPEECH ENVELOPE RATES

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Cochlear Implants (CIs) have provided over 200,000 recipients the ability to perceive sound, but the ability of recipients to understand speech in difficult listening environments varies widely. Speech can be characterised by the envelope and fine structure, CIs do not transmit any fine structure information. Speech understanding for CI recipients is based mainly on the transmission of low frequency amplitude modulations across multiple stimulation channels. In normal hearing listeners amplitude modulated auditory steady-state responses increase neuronal activity in different brain regions from brainstem to cortex with decreasing modulation frequency.

We aim to investigate the neural generators of electrically evoked auditory steady-state responses (EASSR) modulated at frequencies dominant in the speech envelope. We determine the effect of modulation frequency on the location and strength of neural generators and compare these results to those from normal hearing listeners. We also look at inter subject variability in generator location and strength. This may possibly lead to objective neural markers for the evaluation of CI speech processing algorithms and study of post implant neural plasticity.

EASSRs to amplitude modulated pulse trains have been recorded in 5 users of the Cochlear Nucleus CI using a 64 channel EEG system. Pulses were presented in bipolar mode at 900 pps and amplitude modulated at 4,10, 20, 40 and 90 Hz. Source analysis is done using both parametric and nonparametric methods.

Source analysis methods have been validated in normal hearing participants. EASSR responses are detected and initial results have successfully localised the neural generators of amplitude modulated stimuli. A complete analysis of all modulation frequencies will be presented at the conference.

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M38: ASSESSING THE EFFECTS OF DEAFNESS ON CORTICAL NETWORKS IN THE DEVELOPING HUMAN AUDITORY SYSTEM

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Deafness in early life disrupts normal auditory development. The effects are likely to depend on etiology and onset of deafness resulting in reported variability in cortical responses to initial cochlear implant stimulation. We examined responses of the developing auditory cortex over the first 6-9 months of cochlear implant use by mapping the evoked cortical neural network. This was achieved by frequency filtering cochlear implant evoked responses in the electroencephalogram and measuring source dipoles using linear constraint minimum variance (LCMV) beamforming.

A group of 18 implanted children was tested, consisting of 10 children receiving bilateral implants simultaneously and 8 who received unilateral implants. The etiology of deafness varied. Trains of biphasic electric pulses were delivered at 250 pulses/s for 36 ms at a rate of 1 Hz from the apical end (electrode #20) of the implanted electrode array. Cortical responses were recorded from 64 cephalic locations following 1 week, 1 month, 3 months, 6 months and 9 months of CI use. The evoked signals were analysed in 1 Hz frequency bins over a frequency range of 0.5-70 Hz. An LCMV beamformer was used to locate and measure evoked dipoles of activity in the brain. In general, beamformers use spatial filtering to separate signals that overlap in frequencies but originate from different sources. The LCMV beamformer maximally suppresses activity contributions from regions other than that of the desired dipole and, as previously reported, can successfully suppress stimulus artifact from the cochlear implant.

Analyses were restricted to dipoles which were significantly larger in the post-stimulus interval than in the 200 to 80 ms baseline activity recorded prior to stimulus onset. The significant dipoles in the evoked response from 50-800 ms were parsed into well characterised frequencies: delta-band activity (δ): 0.5-4 Hz, theta-band activity (θ): 4-8 Hz, alpha-band activity (α): 8-13 Hz, beta-band activity (β): 13-30 Hz and gamma-band activity (ϒ): 30-70 Hz. These wavelength separations allowed us to evaluate specific oscillations of different cortical processes and to further investigate the spatial and temporal resolution of each.

Density plots of dipoles in each frequency band were created. For all frequency bands, there appeared to be one main response which varied in latency between children. We found that delta waves, typically associated with synchronous spike activity, occurred very early in the response, approximately 50-100 ms post stimulus. Both theta and alpha waves showed oscillations of activity between 100-550 ms. The main activity occurred around 100-200ms post stimulus and decreased over time. This decay was slower for alpha than theta responses. Beta waves were also rhythmic but at a faster rate and with a slower decay. Gamma waves, associated with conscious cognitive processing, did not show significant time-locked activity. Results confirm that a complex network of cortical activity is evoked by cochlear implant stimulation in children who are deaf. Differences from normal are being assessed in follow up studies with an emphasis on understanding the impacts of etiology of deafness.

M39: MECHANISMS UNDERLYING HEARING PRESERVATION IN CI USERS

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With the development of a new soft surgical technique it has been proposed that deeper electrode insertion into the apical region of the cochlea is possible without posing a great threat to residual hearing. With increasing use of this technique, it is important to understand whether this practice is in fact augmenting better speech understanding. Recent studies have demonstrated that low-frequency hearing preservation (auditory thresholds) is possible with implanted patients and studies are beginning to reflect benefit in clinical outcomes. However, the neural mechanisms underlying this benefit are unknown.

Auditory evoked potentials reflect neural activity related to stimulus processing in the auditory cortex. In particular, the N1-P2 complex (responses at 50-200ms after stimulus onset) and the Acoustic Change Complex (ACC) are sensitive to changes in acoustic features such as the spectral and temporal properties of the stimulus.

Sensitivity to stimulus pulse phase duration (ppd) has been shown to decrease in animals with increased nerve damage. However, it is unknown whether this decrease in sensitivity is reflected at the level of the auditory cortex. The objective of this study is to determine mechanisms underlying CI listeners' better performance with hearing preservation, by measuring behavioral and cortical sensitivity to stimuli.

We compared sentence recognition in quiet and in noise, the psychophysical just noticeable difference (jnd) for ppd, and the ACC response evoked by the just noticeably different stimuli in CI users with and without hearing preservation.

Preliminary results suggest that CI users with hearing preservation have better sentence scores in noise and better ppd jnds than CI users without. Latencies and amplitudes of the ACC N1 and P2 were decreased and increased, respectively, for the CI users with hearing preservation. Ongoing studies are also examining psychophysical, ECAP, and cortical responses to envelope changes in the same patients.

Results will help us to discover the effect of the input at the level of the auditory nerve, as well as at the level of the auditory cortex. Cortical responses should also reflect top-down processing. Therefore, these studies will help us to separate auditory nerve input from top-down processing and begin to determine how these affect performance individually.

M40: EXPANDING COCHLEAR IMPLANT NEURAL RESPONSE MEASUREMENT CAPABILITIES TO CORTICAL POTENTIALS

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The recording capabilities of a standard cochlear implant (CI) allow the routine measurement of neural responses from the auditory nerve (ECAPs). While ECAPs can be clinically useful for estimating threshold levels, they are not a good predictor of higher level outcomes such as speech perception. Recently research by our group has shown how the recording capabilities of a standard CI can be expanded to capture cortical neural responses [1]. In a separate study, we showed how CI subject performance in a complex psychoacoustic task, such as spectral ripple detection, can be objectively estimated using cortical responses recorded with standard scalp electrode techniques [2]. Here, we expand the CI cortical potential recording method by validating it in a wider subject population and explore methods from integrating it with more complex evoked potential paradigms that can accurately predict performance on psychophysical tasks.

To record cortical evoked potentials using a CI one intra- and one extra-cochlear electrode are set as the active and reference recording electrodes. Cortical potentials are typically low frequency (<35 Hz) and are a few hundred milliseconds in duration. However, the hardware on a standard CI is optimized for recording ECAPs, i.e. higher frequency neural responses, lasting only a few microseconds. Therefore, several recording buffers must be patched together to sample a cortical waveform, making the recording process time consuming. A second limitation is that when the CI is in recording mode it is difficult to present the type of complex stimuli which can fully characterize a subjects performance. In spite of these hardware and software limitations, we find that it is feasible to record cortical neural responses to complex stimuli using a standard CI.

Combining the CI cortical potential recording method with complex stimuli and evoked potential paradigms may provide an objective, clinically relevant, method for assessing CI user's performance. Such objective metrics are of particular interest for new CI users who have yet to develop speech perception or in pediatric users who are unable to participate in standard psychophysical or speech perception tests. The application and study of these methods would be greatly aided by a new generation of CIs with improved cortical potential recording capabilities such as longer recording buffers, lower sampling rates or optimally positioned extra-cochlear electrodes.

M41: INVESTIGATING A NOVEL METHOD FOR BILATERAL COCHLEAR IMPLANT FITTING: SPEECH OUTCOMES AND ENHANCED ILD DISCRIMINATION

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Traditional approaches to bilateral cochlear implant fitting treat each ear separately in order to optimize hearing in the unilateral condition before putting the two ears together. One potential concern with this approach is that asymmetries across the ears may introduce perceptual distortions to interaural level differences (ILDs), which bilateral cochlear implant listeners rely on for localizing sounds in the horizontal plane. We propose and describe an alternative bilateral fitting method that integrates loudness and centering judgments during simultaneous stimulation of the ears to set clinical stimulation levels (MAPs).

Bilateral research and traditional fitting methods were used and compared in a group of bilateral Nucleus CI subjects. A research software platform allowed adjustments to stimulation levels across the ears, while the commercial fitting software made independent ear measurements. Each fitting method yielded a set of stimulation levels for the tested MAPs. Speech reception thresholds, in the presence of 4-talker babble, were measured for bilateral and unilateral listening conditions and for both MAPs in each subject. ILD discrimination thresholds were also measured using direct audio input with speech stimuli. Results show that speech reception thresholds were unchanged for the different fitting methods, but that ILD discrimination was significantly enhanced by the bilateral research fitting method.

M42: MODELING SOUND LOCALIZATION IN COCHLEAR IMPLANTS

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We have developed a model framework for evaluating interaural time differences (ITD) provided by left- and right ear auditory-nerve responses. Together with our large population model of the electrically excited auditory nerve we achieved quantitative comparisons between different cochlear implant coding strategies. As the model of electric excitation model includes effects such as channel crosstalk, neuronal adaptation and mismatch of electrode positions between left and right ears, its predictive power is much higher than an analysis of the electrical impulses delivered to the electrodes alone. Where it is already known that CI users can interpret interaural level differences to localize sound sources, we focus on the evaluation of ITDs in this investigation. In the intact hearing system, ITDs account for the high spatial acuity especially in the low-frequency region of hearing.

For the evaluation we used realistic signals: our test scenario was a spoken sentence and we moved the virtual speaker around the virtual head of a cochlear implant patient. With a HD CIS strategy, the analysis indicated that the precision of the ITD coding was not sufficient for sound localization. However, the evaluation of a fine-structure coding strategy (FS4, MED-EL) revealed that, this strategy should be able to provide ITD cues with sufficient precision for at least coarse sound localization: the temporal precision of the strategy is sufficient to separate seven spatial positions in the horizontal plane. However, the strength of the correlation between left and right ear auditory spike trains is always much weaker (in our investigation by more than one orders of magnitude) than in the normal hearing case. At location between two electrodes the correlation is even weaker due to channel crosstalk.

In summary, the model framework introduced here is a valuable tool for the pre-evaluation and optimization of bilateral cochlear implant coding strategies. It provides quantitative values for the strength of the correlation between left and right ear auditory spike trains. The final evaluation, whether these cues can actually be exploited by cochlear implant users has yet to be determined by listening tests.

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M43: AGE AT ONSET OF HEARING LOSS, BILATERAL COCHLEAR IMPLANT EXPERIENCE AND SPATIAL RELEASE FROM MASKING

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Bilateral cochlear implantation is widely becoming a standard of care in many clinics for patients who have profound hearing loss in both ears, and improvements over unilateral implant use have been demonstrated in sound localization ability and speech understanding in noisy environments. Despite these improvements, there is high variability in performance among bilateral cochlear implant (CI) users. This variability may arise due to factors such as the patient's hearing history, degraded neural pathways following periods of auditory deprivation, differences in the interaural anatomical placement of the electrode arrays in the two cochleae, etc. One measure of speech-in-noise performance that has shown a high variability among CI users is the benefit of spatial release from masking (SRM), which is the improvement in the speech reception threshold (SRT) when target and maskers are spatially separated.

In this study, we investigated whether there is a relationship between patients' hearing history, years of CI experience, and their ability to obtain a benefit from SRM. SRTs were measured using a method of constant stimuli for co-located and spatially-separated target and maskers in a sound-attenuated booth using the patients' clinical speech processors. The target was a single CNC word spoken by a male talker and the maskers were IEEE sentences spoken by a female talker. In the co-located condition, both target and maskers were presented from the same loudspeaker in front of the subject. In the spatially-separated condition, the target was presented from the front loudspeaker, and a different masker sentence was presented from each of the left and right loudspeakers. SRM was calculated as the difference in SRTs obtained between the spatially-separated and co-located conditions.

To date, data has been collected from 15 adult subjects, aged between 32-73 years old, with 2-15 years of unilateral (and 0.5-9 years of bilateral) CI experience. All were implanted as adults. Reported age at onset of hearing loss ranged from 2-53 years old. SRM benefit ranged from 0-9 dB. A linear model fit to the data ($R^2 = 0.9$) found that years of bilateral CI experience, age at testing, age at onset of hearing loss and the proportion of life with hearing loss were all significant factors affecting SRM performance. Predictions made from the model showed that with later onset of hearing loss, SRM benefits were greatest immediately after implantation but performance steadily decreased with time. With earlier onset of deafness, the predicted SRM benefits were smaller but more stable over time. These results suggest that the high variability in SRM benefit among bilateral CI users may be related to their age at the onset of deafness and years of bilateral CI experience.

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M44: EMERGENCE OF SPATIAL HEARING ACUITY IN YOUNG BILATERALLY IMPLANTED CHILDREN

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Bilateral cochlear implants (BiCIs) are being provided to infants and children at a growing rate. A recent study (Grieco-Calub&Litovsky, 2012, Ear Hear. 33: 1-12.), measured spatial acuity in toddlers fitted with unilateral vs. BiCIs. Some toddlers with BiCIs were able to discriminate reliably the location of a sound source, with minimum audible angle (MAA) thresholds in the range of thresholds measured in normal-hearing age-matched peers. In contrast, other toddlers with BiCIs, and all toddlers with unilateral devices were unable to perform the task even at large angular separations between source locations. The large variability in outcome on this task has spurred interest in the emergence of spatial acuity in these children as they grow older and gain more bilateral experience. The current investigation focused on 12 of the toddlers who returned for testing when they were between the ages of 4-7 years. All 12 children had BiCIs at the time of testing: 4 had received a second implant since the previous study and 8 had gained more bilateral experience. We tested the hypothesis that early bilateral stimulation is not required for emergence of spatial hearing acuity.

Children were re-tested on right-left discrimination, and at this older age they were also tested on a sound source localization task. Stimuli were speech, presented at 60±4dB SPL. Feedback was provided following each response via a computer monitor and a flashing picture corresponding to the correct response.

Results suggest that there is an emergence of spatial hearing acuity not only in the group with early bilateral implantation, but also in children who receive their 2nd implant between 2.5 to 3.5 years of age. In fact, some of the best performers were children who were unable to perform the right-left task as toddlers (from both the bilateral and unilateral groups). MAA thresholds were <18° for 10/11 children with ≥36mo bilateral experience, regardless of their age at the time each received their 2nd implant (14-43 mo). Root-mean-square (RMS) error for sound localization decreased (improved) within subjects with additional bilateral experience (for subjects who were tested longitudinally). However, overall analysis of RMS error revealed no correlation with number of months of bilateral listening experience, MAA, or age at the time of 2nd CI activation.

These findings suggest that early bilateral implantation may not be required for the emergence of spatial hearing acuity, as measured with the MAA task. Further, on behavioral measures such as those used here, children can change their “performance profile” between the ages of toddlerhood to childhood (over a 3-5 year period), thus, early assessment is not predictive of later abilities.

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M45: PSYCHOACOUSTICAL AND SELF-REPORTED OUTCOMES OF A CO-ORDINATED BILATERAL COCHLEAR IMPLANT SYSTEM IN ADULT USERS

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Most bilateral cochlear implant (CI) users have sound localisation and speech recognition in noise that is better than unilateral CI users and worse than normal-hearing people. The discrepancy between bilateral CI users and normal-hearing people partly represents technical limitations (e.g. the signal processing), which hinders the use of binaural cues (i.e. interaural time difference, ITD, and interaural level difference, ILD). For example, independent and unsynchronised signal processing in the two CIs might distort binaural cues. The Digisonic® SP Binaural (DSPB) aims to improve binaural hearing further by providing coordinated processing of the acoustical input. Unlike conventional bilateral CIs, the DSPB implant is equipped with a single speech processor and two microphones, allowing both cochleae to be electrically stimulated.

The purpose of our study is to: 1) assess the DSPB in adult users in terms of sound localisation, speech recognition in noise and self-report measures and 2) evaluate the relative contribution of ITD and ILD cues in localisation and speech recognition.

So far, six postlingually deafened adult users have been tested. Localisation ability was assessed using speech, broadband noise, high-pass noise (ILD dominant), low-pass noise (ongoing ITD dominant) and transient noise (envelope ITD dominant). Each stimulus was presented through one of 11 loudspeakers, selected randomly, extending from -90° to +90° azimuth, repeated six times. Speech recognition thresholds were gathered using the triplet digit test when the speech and noise were spatially co-located and separated (±90º azimuth) to determine the spatial release from masking (SRM) and squelch, head shadow and summation effects. The performance in everyday listening environment was explored through the Speech, Spatial and Qualities of Hearing Scale (SSQ).

The localisation results showed that the root mean square (RMS) error averaged across all the six stimuli was better than expected by unbiased guessing in all cases. There is a trend for RMS error to be poorer with low-pass noise than with the other stimuli. Five of the six subjects show signs of a SRM; the other components and the SSQ data are currently being analysed. Although this experiment is ongoing, the results so far are broadly consistent with the existing literature on bilateral CIs.

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M46: LOCALIZATION OF CONCURRENT SOUND SOURCES USING BILATERAL COCHLEAR IMPLANTS: INFORMATIONAL AND ENERGETIC CONSIDERATIONS

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The ability to perceive, locate and understand target signals is often more difficult in noisy settings due to a combination of energetic masking (EM) that results from spectral overlap between concurrent signals at the auditory periphery, and informational masking (IM) that is caused by confusion at a higher central level due to similarity between the competing signals and uncertainty in how to group signal components (streaming). While perceived location can facilitate correct streaming, localization itself is likely to be affected by both EM and IM considerations. EM may degrade binaural cues both in magnitude and reliability, and IM may make it more difficult to separate signals perceptually and hence assign unique locations to them, even if EM is absent. The present study aimed to assess the effects of EM and IM on the ability of bilateral cochlear implant (BiCI) to localize concurrent talkers.

Eight BiCI users were tested using a concurrent sound source direction identification task. On each trial, two signals were presented simultaneously using an array of loudspeakers spanning the frontal 180° arc, and listeners were asked to indicate the loudspeakers from which S1 and S2 originated. Although the array contained 9 equidistant loudspeakers in total, listeners were unaware of the fact that only 5 were active, and the remaining 4 served to reduce response quantization. One of the two signals (S1) was a recorded sentence spoken by a male talker (M). The other (S2) was varied across conditions and included a recorded sentence from a female talker (F), a modified version of F in which each token was time reversed (F(-t)) and speech-modulated noise (N) with an average spectrum that matched that of F. Tokens in the S2 signal were also either synchronised (sync) or interleaved (inter) with those of S1. It was assumed that interleaving the two signals eliminated EM but also presumably reduced IM to a degree, and that varying the content of S2 from F to F(-t) to N progressively reduced IM without significantly altering average EM. Baseline localization ability was also measured with S1 alone.

Participants performed most poorly in the M+F sync condition, for which EM and IM were both high. Best performance was seen for the interleaved conditions for which EM is absent, but IM may also be significantly reduced. Intermediate outcomes were observed for the M+N sync and M+F(-t) sync conditions which suggests a reduced impact of IM on localization in these signals compared to the M+F condition. These findings suggest that both EM and IM impair the ability of BICI users to locate sound sources in noisy settings.

M47: SEQUENTIAL STREAM SEGREGATION IN BILATERAL COCHLEAR IMPLANT USERS

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Auditory stream segregation is the process whereby a complex sound mixture is broken down into auditory components that are then grouped into perceptual auditory streams, ideally reflecting each individual sound source in the listening environment. Auditory components with similar properties tend to be grouped into the same stream while those with conflicting properties are segregated into different streams. Interaural differences, which for example vary with sound source location, provide one such set of properties, and are more effective in facilitating stream segregation when auditory components follow each other in time, referred to as sequential stream segregation, in contrast to when they occur simultaneously. The present study aimed to psychophysically explore the potential of interaural differences for promoting sequential stream segregation in bilateral cochlear implant (BiCI) listeners.

Six BiCI users were tested using a rhythm irregularity detection task. Stimuli were sequences of alternating ‘A’ and ‘B’ bursts, with either a ‘regular’ uniform timing structure, or an ‘irregular’ structure that was generated by adding a delay to the onsets of the ‘B’ bursts without adjusting onset times of the ‘A’ bursts. Stimuli were presented in 2-AFC task in which listeners were asked to indicate the interval with the irregular rhythm sequence. 'B' burst delay thresholds which enabled rhythm irregularity detection were determined using an adaptive 2-down, 1-up procedure that targeted 70.7% correct performance. It was assumed that the task would be easier when ‘A’ and ‘B’ bursts were perceived as a single stream, and conversely that high thresholds would suggest that ‘A’ and ‘B’ bursts were segregated into different streams.

Burst ‘A’ was presented monaurally in the left ear at a fixed mid-array electrode. The 'ear' of stimulation for burst B was varied to include ipsilateral, contralateral, and diotic conditions. The electrode selected for burst ‘B’ was varied to produce place separations of 0, ±4, and ±7 electrodes relative to the ‘A’ electrode. Results averaged across subjects showed that thresholds for all three ear conditions were significantly different, and were lowest for the ipsilateral condition and highest for contralateral case. For the ipsilateral condition, average delay thresholds also increased as electrode separation increased, which agrees with results from previous studies. The results from the present study indicate that differences both across and within ears can be used to promote sequential separation of multiple auditory streams in BiCI users.

M48: BENEFIT OF CONTRALATERAL ROUTING OF SIGNALS FOR UNILATERAL CI USERS

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For both hearing impaired (HI) and cochlear implant (CI) users, understanding speech in noise remains a challenge. One way to improve speech understanding in noise for HI with a unilateral hearing loss is by Contralateral Routing of Signals (CROS). Such a CROS-system captures sounds with an additional microphone at the worst ear and transmits these to the best ear. The best ear is then provided with a mix of signals originating from both ears. Although the properties of a CROS-system are well-known for subjects with a unilateral hearing loss, these are less well quantified for CI-users. Therefore this study investigates the benefit of such a system for unilaterally implanted CI-users.

Five CI-users are tested with and without the CROS-system. Intelligibility experiments are performed in diffuse noise, where the speech direction was either from the left, front or the right side. Results show that the CROS-system gave an average 8.4 dB improvement in speech reception threshold (SRT) when the speech originated from the CROS-side. An average 4.4 dB loss was measured when speech was directed towards the implanted side. No significant effect was measured for speech from the front.

It turns out that the SRTs from this experiment with CI-users can be accurately predicted (RMSE of 1.1 dB) by the Speech Intelligibility Index (SII), despite the fact that the SII was originally set up to simulate the performance of a unilateral hearing-loss. Except for an overall decrease in intelligibility, no additional modifications were required to let the SII correctly predict the results for CI-users. Supplementary conditions are evaluated with the SII including speech from all directions and directional noise instead of diffuse noise. In contrast to diffuse noise, these results show that with directional noise the effect of the CROS-system is symmetric, with a maximum loss/gain in SRT of 10 dB. In addition, the difference in SRT between maximum and minimum performance due to signal direction and noise type is much smaller with the CROS-system (5.5 dB) than with no CROS-system (20.5 dB).

It can be concluded that in diffuse noise the increase in speech understanding due to a CROS-system is larger than its loss. With directional noise even larger gains in SRT can be achieved. However, in the latter case the effect is symmetric and a similar loss in SRT may occur depending on speech and noise direction. When the user has additional activation control of the system, this disadvantage may be compensated for.

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M49: WIRELESS MULTI-MICROPHONE TECHNOLOGY AND ITS ADVANTAGES FOR BILATERAL COCHLEAR IMPLANTS USERS

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Introduction

Many cochlear implant (CI) users are able to understand speech in quiet very well. Hearing in noise however, remains a challenge. For combating noise there are two main technical approaches: beamformers and (single-channel) noise reduction algorithms. In the hearing aid field beamforming techniques have become more sophisticated with the introduction of wireless communication between bilateral devices. With this wireless signal transmission, the directivity of a beamformer can be increased. Furthermore, the disadvantage of head shadow, when speech comes from the contra-lateral side, can be overcome. Based on a study with unilateral CI users, the benefit of this wireless multi-microphone technology was investigated for bilateral CI users.

Study design

Eleven experienced CI users participated in this study. All were bilaterally implanted with Advanced Bionics HiRes90k implants and used the Harmony sound processor. Two state-of-the-art hearing aids were used that communicated with each other via a wireless link. Both hearing aids were modified so that the analog output could be fed into the external input of the CI sound processor. In a circle of eight loudspeakers the speech reception threshold (SRT) was measured for different conditions using the Oldenburg sentence test (OlSa). Stationary, uncorrelated noise was presented from all loudspeakers with an overall level of 65 dB SPL. Speech was either presented from the front (0°), from 45° or from 90° to the right of the listener. The following microphone settings were tested: a) omni-directional, b) monaural adaptive beamformer, c) wireless binaural beamformer and d) wireless manual directivity steering. In addition, the ClearVoice noise reduction algorithm was used.

Results

The largest group mean benefit in SRT was 4.6 dB, found for the binaural beamformer. This was for speech from the front. For speech from 45°, the adaptive beamformer showed the largest benefit of 2.3 dB. For speech from 90° the wireless directivity steering provided the largest benefit of 1.1 dB. All comparisons were with respect to the omni-directional microphone. The ClearVoice algorithm added another 0.7 to 1.2 dB of benefit, depending on the condition tested.

Conclusion

Beamforming technology based on wireless communication between the two speech processors can improve speech intelligibility for bilateral CI users. In the future, this technology may also be used for bimodal patients (CI + HA).

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M50: IMPROVING COCHLEAR IMPLANT PATIENTS' SPECTRAL RESOLUTION BY DIVIDING THE SIGNAL ACROSS EARS

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Cochlear implant patients' performance is greatly limited by poor spectral resolution, likely resulting from interactions between adjacent electrodes. One potential method for reducing those interactions and thus improving performance is to divide adjacent spectral channels across ears, an approach referred to here as interleaved processing. This reduces the number of stimulation sites per ear, and increases the distance between stimulation sites, while maintaining all the spectral information when the signal from each ear is combined. The benefit of interleaved processing depends strongly on the degree to which this approach reduces the interaction between electrodes and thus increases the resulting spectral resolution.

Although the current fields in opposite ears will not directly interact with each other, they can potentially mask each other perceptually when neural activity from each ear is combined in the central auditory system. This could potentially reduce or eliminate the benefit of interleaved processing. Ipsilateral and contralateral masking was measured to determine the extent to which this occurred. The results indicated that the masking function was consistently narrower when the target and masker were in opposite ears, suggesting that moving some stimulation to the opposite ear can reduce perceptual interactions between channels.

To determine if spectral resolution improved with interleaved processing, patients were tested with a modified version of the spectral ripple task, called the Spectral-temporally Modulated Ripple Test (SMRT). The results indicated that distributing adjacent spectral channels across ears yielded an improvement in spectral resolution.

The results from these experiments suggest that using interleaved processing is a potentially promising method for improving cochlear implant patients' performance, particularly in complex listening environments that require better spectral resolution.

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M51: EFFECT OF PHONOLOGICAL ERRORS ON SPEECH INTELLIGIBILITY AND DURATION OF SENTENCES OF COCHLEAR IMPLANT USERS

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Cochlear implants (CI) allow children with hearing loss to achieve speech perception and production outcomes that make their speech understandable to normal hearing adult listeners. This capability is characterized by a wide variability of scores. In order to understand the factors that contribute to the overall variability, we investigated the effects of sentence duration, contextual information and phonological errors on speech intelligibility and sentence duration over time. Participants were 107 children implanted between the ages of 2 and 4 and tested at 2 time points where they were 8 and 16 years old. Participants' speech intelligibility and duration of sentences were tested using McGarr sentences, which vary in length from 3 to 5 to 7 syllables and were defined as low and high context sentences. Recordings were analyzed using acoustic software to designate the beginning and end of each sentence. Three normal hearing listeners heard, independently, one sentence from one child and wrote down the words they understood. The average score was calculated. Result showed significant negative correlation between speech intelligibility scores and duration of each sentence, effect of context and effect of phonological errors on speech intelligibility. In addition, continued experience with a CI improves intelligibility.

M52: LONG TERM RESULTS OF ABI IN CHILDREN

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Auditory brainstem implants (ABI) can provide useful auditory perception and language development in deaf children who are not able to use a cochlear implant (CI). We prospectively followed-up a consecutive group of 64 deaf children up to 12 years following ABI implantation. The etiology of deafness in these children was: cochlear nerve aplasia in 49, auditory neuropathy in 1, cochlear malformations in 8, bilateral cochlear post-meningitic ossification in 3, NF2 in 2, and bilateral cochlear fractures due to a head injury in 1. Twenty nine children had other congenital non-auditory disabilities. The follow-up ranged from one to 12 years. Children were fitted with a Cochlear 24 ABI device or the MedEl ABI device and followed the same rehabilitation programme. Perceptual auditory abilities were evaluated with the Category of Auditory Performance (CAP). No child was lost to follow-up and there were no exclusions from the study. All children showed significant improvement in the auditory perception with implant experience. Seven children (11%) were able to achieve the highest score on the CAP test; they were able to converse on the telephone within 3 years of implantation. One third of all children achieved open set speech recognition (5 or greater CAP score). All children showed continued improvements in auditory skills over time. The long-term results of ABI implantation reveal significant auditory benefit in most children, and open set auditory recognition in a few.

M53: THE DILEMMA OF CND IN CHILDREN : CI OR ABI? PERSONAL EXPERIENCE ON 21 CHILDREN FITTED FIRST WITH CI AND SUBSEQUENTLY WITH ABI

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Objectives: To compare perceptual auditory abilities in 21 children with suspected cochlear nerve deficiency (CND) and surgically verified absent cochlear nerve absence (ACN), receiving first a cochlear (CI) and subsequently an auditory brainstem implant (ABI).

Study Design: Retrospective cohort study from 2000 to 2011.

Population: Between 2000 and 2011, 21 children initially received a CI at outside institutions and failed to progress in perceptual auditory abilities.

Methods: Explantation of the CI and ABI placement on the same side. All surgical videos were reviewed to determine presence or absence of the cochlear nerve. Perceptual auditory abilities obtained after the CI and after the ABI were converted to the Category of Auditory Performance (CAP) scale.

Results: Preoperatively, all children had severe to profound SNHL and CND. MRI documented ACN in 13 and small cochlear nerve (SCN) in 8 children. The internal auditory canal (IAC) was normal in 6, atretic in 5 and of reduced diameter in the remaining 10 children The vestibular nerve was present on 8 children. At surgery, all patients demonstrated absence of the cochlear nerve. With cochlear implants, all patients had a CAP less than or equal to 2 (0 in 11 pts, 1 in 7 pts, and 2 in 3 pts). With ABI, All patients had a CAP greater than or equal to 2 (2 in 3 pts, 3 in 5 pts, 4 in 5 pts, 6 in 6 pts, and 7 in 2 pts). The complication rate was similar between CI and ABI patients.

Conclusions: Children with a clinical diagnosis of CND and imaging of ACN and SCN, fitted with CIs have an extremely poor prognosis to develop useful auditory abilities. In these children the opportunity for developing open set speech perception and acquiring speech may be better achieved with ABI placement. The overall complication rate of ABI is not greater than that of CI. These results have extremely important implications in the choice of auditory implant in patients with CND and suggest that ABI may be preferable in CND.

M54: NARRATIVE LANGUAGE SKILLS IN SCHOOL-AGED CHILDREN WITH COCHLEAR IMPLANTS

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Introduction: Narratives are a comprehensive measure of spoken language. They not only provide information on discourse but also on component skills such as semantics, syntax, working memory and general knowledge base. Furthermore, narrative skills are strong predictors of written language, reading comprehension, academically related language skills and participation in mainstream education. The study aimed at evaluating narrative skills of a contemporary cross-section of profoundly deaf school-aged children using cochlear implants in order to gain insight into narrative sub-skills in relation to diminished auditory input. Additionally, the study aimed at assessing narrative skills of a high potential subgroup that was selected on four criteria: (1) early cochlear implantation, (2) bilateral cochlear implants, (3) no additional disabilities and (4) one spoken language.

Methods: This prospective multicenter study included 66 prelingually deaf, school-aged children with normal intellectual abilities who received a cochlear implant before the age of five and 66 matched normal hearing peers. Their narrative skills were evaluated by means of the Bus Story subtest of the Renfrew Language Scales. This standardized retelling task was video recorded to allow precise and detailed macro- and micro-level scoring. At the macro-level global organization of content was measured. At the micro-level, a more local level of language use, within- and across-sentence structures were investigated.

Results: The total group of children with cochlear implants demonstrated good results on quantity and coherence of the utterances ($p > .05$), but problematic outcomes on quality, content and efficiency of the retold stories ($p < .05$). The results of the high potential subgroup hardly differed from the results of their age matched peers. Although the high potential CI group produced more incorrect utterances ($p < .05$), their retold stories were not significantly different compared to the control group on any other characteristic ($p > .05$).

Conclusions: Comparison of narrative skills of a contemporary cross-section of school-aged children using CIs with a matched group of normal hearing peers demonstrated good results on quantity and coherence of the utterances, but problematic outcomes on quality, content and efficiency of the retold stories. However, a subgroup of children without additional disabilities who received their CI before the age of 2yr, used bilateral CIs, and were raised with one spoken language did not differ from matched peers except for the percentage of incorrect utterances. This good outcome, especially on the macro-level, implies excellent prospects on reading and writing development and educational outcome, but adequate intervention and follow-up remain essential.

M55: CHARACTERISING CI CHANNEL PERFORMANCE USING AUDITORY STEADY-STATE RESPONSES

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Cochlear Implants (CIs) allow deaf people to perceive sound and enable the average recipient to conduct a conversation in quiet with relative ease. To obtain optimal results CI devices must be individually programmed by defining threshold and comfort levels for each channel which requires a trained clinician, time and a cooperative recipient. Objectively characterising each channel would allow for more accurate fitting of infants, uncooperative recipients and save on overall fitting time.

We aim to develop objective methods for generating CI mapping parameters based on electrically evoked auditory steady-state responses (EASSR), specifically EASSR modulation frequencies dominant in the speech envelope. In this investigation we will determine the influence of CI stimulation electrode and speech envelope modulation frequency on the EASSR amplitude, phase and signal to noise ratio.

EASSRs to amplitude modulated pulse trains have been recorded in 5 users of Cochlear Nucleus CI using a BioSemi 64 channel scalp electrode system. Pulses were presented in bipolar mode at 900 pps across three separate stimulation electrode pairs and amplitude modulated at 4,10, 20, 40 and 90 Hz. The influence of CI stimulation electrode on the EASSR amplitude, phase and signal to noise ratio of the EASSR are investigated.

EASSR responses are detected and show variation across CI stimulation sites. At the conference a thorough analysis and comparison of the across CI stimulation site EASSR parameters we will presented.

This research was supported by IWT-Vlaanderen project 110722 and by the EU within the Marie Curie ITN ICanHear agreement.

M56: MASKING RELEASE IN SPEECH WITH DEGRADED SPECTRAL CUES: EVIDENCE FROM EVENT-RELATED POTENTIALS

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Reduced masking release has been documented in cochlear implant users and normal-hearing listeners when listening to cochlear implant simulations. Among the factors associated with this reduction are the degraded spectral cues available to the listeners. The goal of this study was to explore the neurophysiological responses when listeners are identifying target speech tokens with degraded spectral cues, in steady-state noise and amplitude modulated noise.

We employed the oddball paradigm to measure event-related potentials (ERP) responses for change detection of vocoded speech stimuli in passive and active listening tasks. Synthetic speech tokens of “Wa” and “Ya”, and speech-shaped noise (steady-state or amplitude modulated at 16 Hz) were processed through a 16-band noise vocoder. The standard stimuli in the oddball design were 480 trials of “wa's”, and the deviant stimuli were 120 trials of “ya's”.

Speech tokens were presented at a comfortable level (50 - 60 dB SL) in both quiet and noise conditions. The noise was presented at a level yielding a signal-to-noise level of -3 dB. ERP’s were measured in both the inattentive condition and the attentive condition. In the inattentive condition, the listener’s attention was directed away from the acoustic stimuli by watching a movie; in the attentive condition, listeners were directed to attend to the deviants.

Preliminary data were obtained from three normal-hearing listeners for P300 in the attentive condition when listeners were attending to the deviants in quiet and noise. Results showed progressively increasing latencies (247 ms, 280 ms, and 322 ms, respectively) and decreasing amplitudes (9.3 μv, 4.5 μv, and 3.3 μv, respectively) from quiet to modulated noise and to steady noise. The trend of the ERP amplitude and latency data matched the behavioral results with decreasing D-primes in the task detecting the deviants.

These data indicate: 1. Masking release in normal-hearing listeners was preserved for CV stimuli with spectral cues degraded by means used in the current study. 2. P300 is potentially a good neurophysiological marker for masking release in normal listeners using vocoded“Wa” / “Ya” stimuli. We are currently examining whether the masking release effect would be detected at a pre-attentive level earlier than the P300 response and whether similar results can be obtained from cochlear implant users.

M57: A FLEXIBLE MONOPOLAR STIMULATOR FOR ANIMAL STUDIES IN AUDITORY PROSTHESES

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This paper presents the design of a bench-top monopolar stimulator that can be used for animal studies in cochlear implants. The stimulator is controlled by three high-speed digital Input/Output (I/O) cards manufactured by National Instruments Corporation and is electrically isolated. The stimulator could provide 16 independently controlled charge-balanced monopolar channels, each varying in stimulation parameters. Four stimulation patterns, which are symmetric biphasic, non-symmetric biphasic, triphasic and amplitude modulate biphasic, are available in either simultaneous or interleaved modes. A user-friendly and intuitive Matlab Graphical User Interface (GUI) is provided with the stimulator board to simplify its control and use.

This work supported by contract N01-DC-6-0002 from NIDCD/NIH.

M58: REMOVABLE INSERTION SHANKS FOR PENETRATING AUDITORY IMPLANTS

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Devices implanted directly into the nerves of the auditory pathway (auditory nerve, auditory brain stem, etc.) have the potential to improve hearing restoration and offer options to patients with illness or injury which preclude conventional cochlear implants. Such implants use micro-scale electrode arrays that penetrate into the nerve tissue and these are typically made of silicon [Middlebrooks and Snyder, 2010, J. Neurosci., 30(5), 1937-46; McCreery, 2008, Hearing Res., 242, 64-73]. Flexible polymer electrode arrays are expected to extend interaction with nerves by minimizing adverse tissue response compared to silicon. However, surgical implantation of thin-film polymer electrode arrays remains a challenge because of their flexibility. We have developed several methods of temporarily attaching flexible electrode arrays to silicon stiffeners that can be removed after insertion, leaving the flexible device in place.

In one method, a flexible probe is attached to a stiffener using biodissolvable polyethylene glycol (PEG). After implantation of the stiffened probe assembly, the PEG dissolves, enabling extraction of the silicon stiffener. A novel stiffener design features shallow narrow channels along its length which exploit capillary forces to distribute the PEG adhesive in a controlled manner along the entire stiffener. This technique was used in conjunction with flip-chip bonding to precisely align and attach the probe to the stiffener. In vitro testing in 0.6% agarose gel was used to characterize probe displacement during stiffener extraction, demonstrating displacement of approximately 30 µm. The method has also been used in several in vivo tests in which both local field potential and single spike recordings were successfully obtained from the implanted flexible probe.

In another method, we developed a removable vacuum insertion tool that temporarily attaches to the probe during insertion into neural tissue. The insertion tool is composed of a custom designed vacuum fixture and stiffener that interface to apply suction to the probe. The stiffener designs features an enclosed air-tight channel and vacuum ports through which suction is applied. Upon inserting the probe and placing it precisely within tissue, vacuum pressure is disabled, and the stiffener is extracted.

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M59: CHRONIC MICROELECTRODE SYSTEM FOR PENETRATING AUDITORY IMPLANTS

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Implanting devices directly into the nerves of the auditory pathway provides an option for restoring hearing to patients whose cochlea is damaged by illness or injury. Because these devices are required to function over the lifetime of the patient, there is a need for reliable, long-term prostheses. To fill this need, we have developed a chronic microelectrode system designed for chronic stimulation and recording. The chronic system consists of a flexible polymer array, a removable stiffener, and a specially-designed percutaneous connector.

The chronic microelectrode arrays utilize multiple layers of polyimide and trace metal to fabricate up to 64 electrodes, suitable for either stimulation or recording. The flexible polyimide arrays minimize adverse tissue response, compared to the traditional silicon. The electrodes are comprised of sputtered, thin-film platinum and are on both the top and bottom-sides of the microelectrode array. Both the topside and bottom-side electrodes are recessed from polyimide outer surfaces (i.e. the arrays are axially symmetric) [A. Tooker et al., 34th Conference of the IEEE EMBS, 2012] to generate more uniform electric fields [M. Susserman et al., IEEE TBME, 38(5), 401-408, 2012]. To improve charge storage capacity and increase electrode lifetime, high-surface area platinum is electrochemically deposited on the thin-film platinum after array fabrication. Charge storage capacity and electrode impedance are measured using cyclic voltammetry and electrochemical impedance spectroscopy to characterize electrode performance. Similar microelectrode arrays have been used for chronic in vivo experiments for more than 1 year and have passed the FDA mandated ISO 10993 biocompatibility testing for fully implanted devices.

The microelectrode arrays integrate with both a chronic percutaneous connector and a removable stiffener. A stiffener is required for surgical implantation of the polyimide electrode arrays due to their flexibility. The chronic percutaneous connector has a small form-form factor with a user-friendly low force connect and disconnect feature. The connector can easily be adapted to perform as a fully implantable, hermetically sealed connector. The versatility of the system allows for use in a variety of neural applications.

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M60: CHRONIC PERCUTANEOUS ELECTRICAL CONNECTOR FOR ELECTRICAL RECORDING AND STIMULATION WITH MICROELECTRODE ARRAYS

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The ability to translate fundamental developments in neural stimulation and recording into clinical practice will require the ability to perform long-term animal studies. In recent years, significant advances have been made to extend the lifetime of neural interface technologies. For future clinical application, devices will be fully implanted and wireless, but these devices are not always practical research due to cost, complexity, and inflexibility. There is a growing need for reliable and long-term methods to percutaneously interface neural devices with external electronics. In particular, we are interested in performing chronic experiments for intra-neural auditory prostheses.

State-of-art percutaneous connectors are large, non-biocompatible, require high connection/disconnection forces, and are prone to failure over a limited number of cycles. We have addressed this need by developing a percutaneous chronic connector which is small, utilizes biocompatible components, and requires low connection/disconnection forces. Further, the form factor of this new connector is designed to minimize potential damage due to animal motion and behavior, thereby enabling long-term, chronic studies. Our connector consists of two main components: 1) a percutaneous connector attached to the neural interface and mounted to the animal's skull, and 2) a mating connector attached to external electronics. Electrical contact is made between the percutaneous and external connectors via a silicone-based interconnector with embedded conductive wires that make electrical contact when compressed. A unique magnetic clamping mechanism is used for accurate and passive alignment of the percutaneous and external connectors; it also includes a safety disconnect feature that protects the animal from injury, and the device from damage. Between experiments, the external connector is replaced with a protective cap to keep out moisture and contamination.

Our connector can currently accommodate up to 64 channels with reasonable reliability during electrical testing. The components in contact with the body are assembled of chronically biocompatible materials that have passed ISO 10993 biocompatibility testing standards. We have tested the connector in vitro while attached to a polyimide-based microelectrode array to confirm device performance. When combined with our flexible polymer microelectrode technology, we believe that the device will enable long-term neural stimulation and recording studies that are not feasible using current technologies.

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TUESDAY POSTER ABSTRACTS

T1: WHAT CAN WE EXPECT OF A CONTEMPORARY GROUP OF CHILDREN WITH COCHLEAR IMPLANTS REGARDING THEIR AUDITORY, LINGUISTIC AND COGNITIVE SKILLS?

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Introduction
As a result of neonatal hearing screening and subsequent early cochlear implantation profoundly deaf children have access to important information to master spoken language skills at a young age. Nevertheless, outcome measures still reveal great variability in individual performance. Some children with CI(s) acquire language skills within normal limits while performance of other children remains far below average. Understanding the causes of this variation would allow clinicians to offer better prognoses to CI candidates and efficient follow-up and rehabilitation.

Methods
In order to determine benchmarks with regard to speech and hearing development in young implanted children a multicenter project with a large contemporary group of young participants (age limit=12, n=288) with one or two CI(s) was set up. Five different studies, retrospective and prospective, were carried out, to determine predictors of spoken language development and to investigate benchmarks in auditory performance (compared to normal hearing peers). In addition, the role of neurocognitive factors for spoken language development, auditory skills and reading comprehension was investigated.

Results
In general the different studies show that children implanted under the age of 2 yr, with bilateral CIs, without additional disabilities and using one language yielded age-adequate vocabulary, morphology, syntax and narrative skills. Age at cochlear implantation, contralateral stimulation, additional disabilities, multilingualism, parental involvement and communication mode were strongly related to language performance and accounted for up to 50% of the variance. In expressive language tasks micro-level analyses showed difficulties with morphological and syntactic rules and inefficient narrative skills in comparison with matched normal hearing peers. These linguistic skills are compared to recently acquired neurocognitive and open-set speech in noise data in unilaterally and bilaterally implanted children to investigate the contribution of memory and intelligibility to successful language acquisition and, hence, reading comprehension in children with CIs.

Boons, T., Brokx, J. P. L., Dhooge, I., Frijns, J. H. M., Peeraer, L., Vermeulen, A., Wouters, J., van Wieringen, A. (2012). Predictors of Spoken Language Development Following Pediatric Cochlear Implantation.Ear and hearing, 33(5), 617-639.

Boons, T., Brokx, J. P. L., Frijns, J. H. M., Peeraer, L., Philips, B., Vermeulen, A., Wouters, J., van Wieringen. (2012). Effect of pediatric bilateral cochlear implantation on language development.Archives of pediatrics & adolescent medicine, 166(1), 28-34.

Boons, T., De Raeve, L., Langereis, M., Peeraer, L.,, Wouters, J., van Wieringen, A. . (2013). Expressive vocabulary, morphology, syntax and narrative skills in profoundly deaf children after early cochlear implantation, RIDD, accepted

T2: SPECTRAL RIPPLE DISCRIMINATION IN INFANTS WITH IMMATURE OR ATYPICAL SPECTRAL RESOLUTION

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Spectral ripple discrimination (SRD) has been used to measure spectral resolution (SR) across a broad frequency range in adults with hearing impairment. However, SRD performance in infants may be strongly affected by other factors such as intensity resolution and attention. Previously, we showed that, while overall SRD sensitivity in 7mo normal-hearing (NH) infants was immature, thresholds improved with increasing spectral ripple depth at a similar rate to adults. This finding is consistent with earlier research showing mature SR by 6 months of age. Two experiments were conducted to investigate the effect of spectral ripple depth on SRD sensitivity in participants known to have immature (Experiment 1, 3mo normal-hearing infants) or impoverished (Experiment 2, cochlear implant users) SR. In both cases, the effect of ripple depth on SRD should be significantly smaller than in listeners with normal, mature SR.

Stimuli were broad-band noise bursts filtered by a sinusoidal spectral envelope with logarithmically-spaced peaks. We used the Observer-based Psychoacoustical Procedure to determine the highest ripple density at which a participant could detect a 90º phase shift in the rectified spectral envelope. In Experiment 1 (E1), the effect of ripple depth (intensity) was varied within-subjects, in counterbalanced order, and SRD threshold in ripple density was assessed. Performance was compared with previously reported 7mo data. If 3mo have immature SR relative to 7mo infants, we would expect a smaller effect of ripple depth for the younger infants (Experiment 1). In Experiment 2 (E2), 9 CI participants (3 adults, 5 infants/toddlers, and 1 school age-child) are compared to the NH data. As with the 3mo NH infants, we would expect smaller effects of ripple depth in CI participants compared to NH participants.

To date, results of E1 show no significant difference between SRD performance at low and high ripple depths in 3mo infants, suggesting that the effect of ripple depth is smaller than in 7mo infants. The interaction between age and ripple depth is expected to reach significance when data collection is completed. Results from Experiment 2 show that CI users, on average as well as individually, have smaller ripple depth effects than NH 7mo infants. Again, the interaction between hearing method and ripple depth is likely to reach significance when data collection is complete.

The early findings from E1 and 2 suggest that the effect of ripple depth is lower in NH infants with immature SR and CI users who have impoverished SR. Computing the effect size of ripple depth on SRD sensitivity, therefore, is a promising method for measuring SR in infants independent of intensity resolution and attention.

T3: PITCH PERCEPTION AND LINKS TO LANGUAGE DEVELOPMENT IN PRIMARY SCHOOL CHILDREN WITH COCHLEAR IMPLANTS

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Pitch perception is thought to be a crucial aspect of language development. In child-directed speech, the use of pitch is universal - as well as making a child more likely to be interested in the speech interaction [Jusczyk, Houston, & Newsome, 1999, Cog. Psych., 39, 159-207]. However, children with a cochlear implant may be unable to hear these changes in the pitch due to both the device limitations, and the auditory systems response to the electrical signals from the device [Green, Faulkner & Rosen, 2002, J. Acoust. Soc. Am., 112, 2155 -2164].

In the current literature there is a wide variation in the outcome of children with cochlear implants - both in their ability to perceive pitch and prosodic changes in the auditory signal, and in their developing speech and language. The aim of this research is to establish the range of pitch processing abilities of implanted children and see if this is linked to their speech and language skill. This is based on a view that initial preferences in an infant to hear the prosodic changes in speech aid later language development. Another area this paper aims to address is how a child's ability to hear pitch change is dependent on the context of the sound. Is pitch an independent percept, or do formant changes affect the ability to hear pitch changes?

Cochlear implanted children between 5 and 11 years took part. The first stage of the research focused on the ability of children with a cochlear implant to perceive the acoustic correlates of stress and intonation of speech using a two interval forced choice ranking task with either a harmonic complex or speech-like sounds. The perception of higher fundamental frequency, intensity and duration was measured to determine difference thresholds. Two independent variables were assessed. The first independent variable was the auditory cue presented; fundamental frequency, intensity or duration. The second independent variable - for speech-like sounds - was vowel change with three levels: no vowel change, a change in the first formant of the vowel or a change in the second formant of the vowel. The dependent variable was the participants' threshold on each task. Perception of prosody was also investigated using both natural and manipulated utterances and input components of the PEPS-C task [Peppé & McCann, 2003, Clin. Ling. Phon., 17, 345-354]. Receptive language, speech perception and non-verbal skills were also assessed using a range of standardised tests. Our initial results indicate that there is a great deal of variability in the ability of implanted children, particularly with regard to the perception of pitch and prosodic changes. A link between these acoustic measures and language skill is expected.

Support provided by a Deafness Research UK studentship.

T4: SENSITIVITY TO BINAURAL CUES IN NORMAL HEARING CHILDREN AND CHILDREN WHO USE BILATERAL COCHLEAR IMPLANTS

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Binaural hearing provides a listener with cues that aid in sound localization, namely inter-aural time and inter-aural level differences (ITDs and ILDs). In addition, ITDs in the envelopes of high-frequency carriers that are modulated may also be important. In patients who are deaf, bilateral cochlear implants (BiCIs) are known to lead to improved sound localization abilities compared with single-CI use. Notably, children who use BiCIs still perform significantly poorer than their normal hearing (NH) peers. One reason for this notable gap in performance is due to the limitation in the ability of the speech processors to encode the acoustic signal with fine resolution. The detailed temporal structure of the original sound gets discarded due to the rapidly changing signal, making it difficult to gain access to ITD cues. Additionally, pre-lingually deafened children lack access to acoustic binaural input, thus their auditory system may be insensitive to those cues.

The aim of this study was to investigate the sensitivity of children who use BiCIs to ITDs and ILDs and to compare their abilities to NH children. It is hypothesized that NH children will be sensitive to both ITDs and ILDs, but pre-lingually deafened children who use BiCIs will be sensitive to ILDs only due to the lack of early binaural exposure.

In the lateralization task, subjects were asked to report the perceived intra-cranial position of the stimuli when ITDs and ILDs were imposed. In the discrimination task, a two-alternative forced choice task was used to measure the subject's just noticeable differences (JNDs). Seven children with bilateral Nucleus CIs (ages 9-13) participated in these experiments. Sensitivity to ITDs and ILDs was measured on a single, pitched matched pair of electrodes through direct stimulation with synchronized research processors. Stimuli were 100 pulses per second, constant amplitude pulse trains that were 300ms long. Eleven NH children (ages 8-10) were tested using a 500 ms Gaussian-enveloped tone (GET) pulse train (4 kHz center frequency, 100 pulses per second), presented over headphones.

Results suggest that all children with BiCIs demonstrate sensitivity to ILDs on tasks of lateralization and discrimination. In contrast, ITD sensitivity was measurable only in one post-lingually deafened child. In NH children, with stimuli that mimic implant processing, performance was comparable to NH adults. This suggests that the lack of ITD sensitivity in pre-lingually deafened children who use BiCIs may be due to the lack of access to fine binaural timing information needed during the early years of life and/or neural degradation. Our findings suggest that early exposure to binaural cues may be necessary for achieving best performance.

Work funded by NIH-NIDCD (R01DC8365, Litovsky) and NIH-NICHD (P30HD03352)

T5: COGNITIVE, COMMUNICATIVE AND PERCEPTIVE DEVELOPMENT AFTER THREE YEARS OF IMPLANTATION OF CHILDREN AGED 6 TO 9.

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Studies on the development factors on deaf children with cochlear implant (CI) emphasize the existence of a large inter-individual variability in CI outcomes due to multiple factors. The precocity of the implantation appears to be one of the main factors contributing to the success of rehabilitation and more specifically to the development of their lexical stock. However, the heterogeneity of linguistic performances highlights the fact that the age of implantation is not the only factor that explains inter-individual variability even in cases of early implantation.

The aim of our work is to study the cognitive, communicative and perceptive development of deaf children after a minimum of three years of cochlear implantation to investigate perceptual, cognitive or neuropsychological features that would be related to the level of rehabilitation through the implant. A cohort of profoundly deaf children aged between 6 and 9 was recruited (n=20). The children were assessed with cognitive tests "Memory and Learning" and "Attention and Executive functions" from the NEPSY, Korkman, 1997) and with a test evaluating their perceptive capacity (free categorization task with everyday sounds).

Results showed that in spite of a large variability, CI deaf children present visual (attention and memory) and planning skills which are near normal values or even slightly higher. However, results obtained with an audio-visual memory test (first name memory) or a narrative memory test, showed a tendency in CI children to have performances lower than normal hearing children. Audio-visual memory performance values were correlated with speech intelligibility (evaluated by the Speech Intelligibility Rating, SIR). In addition we also observed a high correlation between performance values in the two memory tests and the lexical and syntactic comprehension scores. Concerning the free categorization task based on environmental sounds, deaf children of 8-9 years old produced classes based mainly on acoustic criteria. On the opposite, age-matched hearing children (n=38) used more often a semantic-based categorization strategy.

Overall, our preliminary data suggests that after three years of implantation, deaf children could have used auditory information, but they also rely on visual canal.

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T6: COCHLEAR IMPLANTATION OF CHILDREN WITH STEEPLY SLOPING HEARING LOSS

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Pre-operative cochlear implant candidacy criteria have expanded to include adults with significant residual hearing but limited speech understanding compared to normal hearing peers. Clinicians and parents continue to be hesitant to implant children with less than severe to profound hearing loss at all frequencies. It is particularly challenging to determine CI candidacy for children with steeply sloping high-frequency hearing losses. This case series describes the cochlear implant candidacy evaluation process and outcomes for four children with significant residual hearing at frequencies < 1 kHz. Pre-implant speech perception testing included age-appropriate monosyllabic words at average (45 dB HL) and/or loud conversational (55 dB HL) levels. These children met the adult criteria for the Nucleus Hybrid with (1) hearing thresholds <60 dB HL at frequencies <500 Hz; (2) thresholds > 80 dB HL for frequencies > 2 kHz; and (3) monosyllabic words scores < 50% in the ear to be implanted and less than 80% contralaterally. Subjects' characteristics included: binaural hearing aids for at least 6 months including use of hearing aids enabled with a frequency compression algorithm for at least 3 months; progressive bilateral sensorineural hearing loss; normal temporal bone anatomy; and participation in auditory-verbal therapy. Despite overall auditory language comprehension skills at or above their chronological age level as measured by standard language tests, parents reported that their children had difficulty understanding speech in everyday situations; and required significant assistance to acquire new information. Parents underwent a psychosocial evaluations by an experienced psychologist; and counseling by both the surgeon and audiologist to ensure that they understood the risk of their child losing all residual hearing in the ear to be implanted. Children ranged in age from 30 to 57 months at the time of unilateral cochlear implantation. Three children received the Nucleus CI422 (SRA) and one the Nucleus 512. At three months post- implant activation each showed a greater than 50% improvement in monosyllabic word understanding at 45 dB HL. One child began to show significant elevation in thresholds at 250 and 500 Hz at 24 months post-implant in both ears with greater elevation in the implanted ear. Two children have shown no changes in residual hearing up to 6 months post-implant, and the other no change at 3 months. These results suggest that children with steeply sloping hearing losses and poor speech perception skills may perform significantly better with a cochlear implant than with hearing aids and should be considered for implantation prior to loss of low frequency hearing.

T7: OBJECTIVE MEASUREMENTS OF BALANCE DYSFUNCTION IN CHILDREN WITH SENSORINEURAL HEARING LOSS AND COCHLEAR IMPLANTS

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We aimed to quantify balance dysfunction in children with sensorineural hearing loss (SNHL) and cochlear implants (CI) using biomechanical measures of stability and movement. We previously used the Balance Subset of the Bruininks Oseretsky Test of Motor Proficiency II (BOT-2) to assess balance dysfunction in children with concomitant cochlear and vestibular end organ dysfunction. While the advantage of the BOT-2 is that it is standardized and has normal scores across a wide age range, it can only capture gross abnormalities in balance and overlooks subtle differences. Moreover, it provides no information about how children move in each task. Differences such as the use of an articulated posture versus en bloc posture could be important in distinguishing both the characteristics and efficacy of the compensatory strategies used by these children for example.

In the present study, we quantified the BOT-2 in the Challenging Environmental Assessment Lab (CEAL). A motion capture system was utilized to quantify the biomechanical properties of the head, trunk and leg segments while children performed each BOT-2 task. Specific measures included: centre of mass (COM), head angles, trunk angles, lower limb speed as well as step length, width and variability. Angles were expressed in degrees from a neutral “zero” position recorded at the start of each trial. Centre of pressure (COP) was also captured using forceplates. The efficacy of the participants postural control strategies was assessed using the scalar distance between COM and COP.

A total of 38 children underwent the test protocol:12 children with combined cochleovestibular loss and bilateral cochlear implants (CI) and 26 normal hearing controls (CONs).The CI group demonstrated poorer performance on the BOT-2 than CONs, consistent with our previous reports. Biomechanical measures revealed abnormally large movements in the CI group in most measures. For example, maximum head roll and trunk pitch angles in the CI group were larger (8.96 ± 2.15° and 5.25 ± 1.59°, respectively) than in CONs (3.26 ± 0.50° and 1.92 ± 0.26°, respectively).

In addition, the CI group used a different postural control strategy as indicated by a larger scalar distance between COM and COP in the y-axis for the CI group (5.9 ± 1.4 cm) than CONs (1.2 ± 0.15 cm) and an increased variability (14.7 ± 4.9 cm in CI versus 1.4 ± 3.4 cm in CONs).

Results indicate that it is feasible to quantify the biomechanical properties of postural control while performing a standardized test of balance function (BOT-2) in children. The sensitivity of the measures reveal the degree of abnormality in balance experienced by children with cochleovestibular deficits who use bilateral CIs and indicate that these children employ different strategies to maintain their balance than their normal hearing peers.

T8: AUTOMATED ECAP CLASSIFICATION

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Electrically-evoked compound action potentials (ECAPs) are an objective measure of cochlear implant stimulation peripheral encoding on the auditory nerve. It is commonly used in clinical routine for assessing the auditory nerve functionality at the end of surgery. Automated ECAP algorithms aim to deliver regular ECAP function outcome 'by single-click' without the need of parameterization of the stimulation and recording settings. As an example, Auto-NRT (implemented in Custom-Sound EP, Cochlear) or Smart-NRI (implemented in RSPOM, AB) automatically measure ECAP thresholds. These automated ECAP-functions decide on which current-unit to set next depending on ECAP classification as containing or not a neural response, until a satisfactory ECAP-threshold is obtained. ECAP classification is a critical step in automated ECAP functions: it should be made as accurate as possible.

Here, a comparison was made between automatic and visual rating of a dataset of 8,375 ECAP traces. These traces were originally obtained with RSPOM 1.3 using Smart-NRI (490 functions, 9969 traces), Recovery (496 functions, 4837 traces) and Spread-of-Excitation (256 functions, 3569 traces). Both criteria retained for automatic ECAP classification was the voltage magnitude of N-P and the signal-to-noise ratio between the section where the neural response was expected and the last 42 samples where neither neural response nor artifact was expected. Altogether, a high degree of correct ECAP classification was achieved using the combination of +5 dB-SNR and 50 µV: it produced the best compromise between high true-positives and -negatives and reasonably low level of false-positives and -negatives. A distinction appeared between Smart-NRI on one side and Recovery / Spread of Excitation on the other side, thought to be due to better quality traces obtained with forward-masking compared to alternate polarity (used in Smart-NRI). These observations are implemented in the automated functionalities of new upgrade of RSPOM: AB-VOLTA.

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T9: PROGRESSIVE LOW FREQUENCY HEARING LOSS IN A GUINEA PIG MODEL OF COCHLEAR IMPLANTATION BY ALTERING ELECTRODE POSITION

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Delayed or progressive hearing loss after cochlear implantation occurs in a small but significant number of human recipients. The causes behind such hearing loss remain unclear, particularly as the phenomenon does not occur in all implant patients. We postulate that the electrode position within the cochlea and the route of insertion (round window/cochleosotmy) may have a role to play in progressive hearing loss. Here we describe a guinea pig model of progressive low frequency hearing loss over three months resulting from round window membrane (RWM) insertion using a muscle plug.

35 guinea pigs had their audiological function tested and then placed into 5 intervention groups; RWM implantation, cochleostomy with medially placed electrode, cochleostomy with a laterally placed electrode, RWM control and cochleostomy control. Auditory brainstem responses (ABRs) and distortion products of otoacoustic emissions (DPOAEs) were used to evaluate auditory function. These were measured pre operatively and again at 1 week, 1 month and 3 months post operatively.

RWM electrode insertion and RWM controls showed a significant progressive hearing loss in the low frequency (2 kHz) region of the cochlea ($p=<0.01$). The absolute change in ABR threshold in the low frequency regions of the cochlea was greater in the RWM control group than in the RWM insertion group and progressive hearing loss was not seen in the cochleostomy groups. The least threshold shift across all frequencies over 3 months was in the lateral electrode cochleostomy group. DPOAE results in the RWM insertion group showed an increased amplitude of response over time but all other groups showed a loss of amplitude including the RWM control group.

In this study, we have shown that a progressive low frequency hearing loss can occur using a normal hearing animal model. As the results from the RWM control group suggest, this hearing loss is not solely attributable to electrode insertion trauma but likely relates to changes to the RWM membrane as result of the muscle plug. The increase in DPOAE amplitude in the RWM insertion group could suggest an electrode effect to the basilar membrane perhaps resulting in basilar membrane hyper mobility apical to the electrode. This result also suggests that hair cell function has been maintained in the RWM insertion group and hair cell death is unlikely to be responsible for the observed changes in threshold across frequencies. Insertion of an electrode through the RWM or muscle plug placement on the RWM, affects low frequency hearing possibly by affecting elements of the cochlea responsible for signal transduction.

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T10: TARGET STRUCTURES FOR INFRARED NEURAL STIMULATION (INS)

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Introduction:
Pulsed mid-infrared lasers have been used as a method for neural stimulation, including cochlear stimulation. The use of lasers is appealing for many reasons, namely the spatial selectivity and non-contact method of stimulation. The mechanism of INS is believed to be involved with local heating of discrete neuron populations. In cochlear INS, some of the heat is absorbed in the cochlear fluid proximal to the stimulation site, resulting in measurable stress relaxation waves. These waves may vibrate the basilar membrane or directly stimulate hair cells, causing a stimulation by-product. The goal of the present study was to determine the contributions of hair cells to the recorded cochlear INS response.

Methods:
In normal hearing animals, auditory responses can be masked with an acoustic stimulus. In the present experiments the response to INS was masked in hearing animals with a broad band noise stimulus. The cochlea was accessed surgically and an optical fiber was inserted through a cochleostomy in the basal turn. After recording the effect of the masking using CAP measurements taken at the round window, the animals were deafened by 25mM neomycin, injected directly in the scala tympani via the cochleostomy. After deafening was confirmed by testing for auditory-evoked CAPs, the masking experiment was repeated. If INS occurs through direct neural interaction, and not from hair cell stimulation, then the masking effect should disappear but the response to INS should change little. Chronic deaf animals lacking hair cells were used in a second series of experiments. Recordings from the central nucleus of the inferior colliculus (ICC) were used to determine whether stimulation of a cochlea lacking hair cells is possible.

Results:
The data shows the masking effect is present in normal hearing animals and disappears after deafening with Neomycin. Animals that showed no hair cells in histologic sections revealed a neural response recorded from the ICC.

Conclusions:
The effects of hair cells on INS can be seen through masking noise over optical stimulation. However, when hair cells are damaged or no longer present, INS still generates a neural response. These findings lead to the conclusion that although INS-induced stress relaxation waves may have some interaction with hair cells, INS occurs through the direct interaction of the radiation and the neuron.

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T11: INVESTIGATING THE EFFECT OF COCHLEAR GEOMETRY AND NEURAL MODELS ON THRESHOLDS WHEN MODELLING THE COCHLEAE OF SPECIFIC IMPLANT USERS

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It is ever the objective of a model to represent the actual system as closely as possible whilst taking into account the required or acceptable accuracy and computational cost. Hence, if the performance of a cochlear implant (CI) is to be optimised for a specific person, a model of that specific person's cochlea and auditory nerve should provide the best platform for the optimisation. The fact that significant variability in effectiveness of CI's in different CI users exist is in itself sufficient motivation for the development of user-specific models. To this end this paper compares the modeled neural excitation profiles of three user-specific volume conduction models of the cochlea to investigate the effect that the cochlear geometry and the choice of auditory nerve fiber (ANF) models, in particular the compartmental model by Rattay et al. (Hear. Res. 2001; 153(1-2):43-63) and the GSEF model by Frijns et al. (Hear. Res. 1995; 87(1-2):170-186) have on neural excitation.

First, three user-specific volume conduction models of three individual implanted cochleae were constructed from in vivo CT scans. Modelling was done similar to the method described by Malherbe et al. (Med. Eng. Phys., DOI 10.1016/ j.medengphy.2012.09.001). Secondly the potential distribution inside each of the cochleae was determined for two scenarios. The first scenario places a generic electrode array in a medial position along the scala tympani of each cochlea isolating the effect of the cochlear geometry. The second scenario replaces the generic array with the user-specific array. The potential distribution results from both scenarios of each person are then used as input to the two ANF models to determine the respective neural excitation profiles for comparison and evaluation.

Further investigative results are presented for the case where the single auditory nerve fibers placed at discrete intervals along the spiral lamina, each representing a cluster of fibers, are replaced by four fibers along the periphery of each cluster's volume inside the spiral lamina. The neural excitation profiles for a basal and an apical cluster, consisting of the four peripheral fibers and single center fiber, are then compared to determine if the four peripheral fibers exhibit a response significantly different to that of the centerfiber. The observations from the comparisons done for each of the three user-specific models are then evaluated to determine if further research into the impact of the ANF density distribution along the spiral lamina is warranted.

T12a: ECAP THREHOLDS IN COCHLEAR IMPLANT USERS MEASURED INTRAOPERATIVELY ARE HIGHER THAN THOSE MEASURED POSTOPERATIVELY

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Measuring the electrically evoked compound action potential (ECAP) is an approach commonly used to check the interface between the electrode contacts of a cochlear implant (CI) and the auditory nerve fibres as well as to find an estimate for the initial fitting parameters of the speech processor. An intraoperative measurement has the advantage that stimulation cannot evoke unpleasantly loud perceptions and is therefore especially suitable for small children. Some research has been conducted already on ECAP thresholds measured at different times after implantation. E. g., van Wermeskerken et al. (Int J Audiol 2006;45:589-594) didn't find a significant difference of ECAP threshold between intraop and postop measurements. This would be most favourable, as the intraoperative values could then be used for postoperative fitting procedures without further verification. However, the development of ECAP thresholds over time is still controversially discussed.

In our study, we analyzed in total 21 patients implanted with a Nucleus CI422 (n=8) or CI24RECA (n=13). We used the CustomSound Software Suite to measure ECAP thresholds via AutoNRT and impedances at three different times: intraoperatively, one to three days later, and approx 5 weeks later.

The ECAP thresholds found intraop were on average 18±11 CL higher than those found one to three days later. After approx 5 weeks ECAP thresholds again came down slightly, on average by 4 CL (±12 CL). No significant changes in electrode impedances were found during that time, fluctuations were smaller than the standard deviation (2 – 3 kOhm).

On one hand, the differences between intraop and postop ECAP thresholds may be caused by the different default measurement parameters used intraop vs. postop, the latter with a lower stimulation rate. On the other hand, Spivak et al. (Ear Hear 2011;32:104-113) found no differences in ECAP thresholds when comparing intraoperative with postoperative outcomes while using the two different parameter sets. The application of an anaesthetic during surgery should not influence ECAP thresholds. Perilymph inside the cochlear could change in it's composition and with it, conductivity might be influenced. Also tissue growth may alter the path of the electric current. To shed more light onto this topic, we also measured spread-of-excitation few days after implantation as well as approx 5 weeks later, additionally to the above mentioned measurements. It seems that there is little change of SOE over time. However, data collection is still ongoing and full results and correlations will be presented at the conference.

T12b: MEASURING ELECTRICALLY EVOKED COMPOUND ACTION POTENTIALS AS A FITTING TOOL FOR ABI PATIENTS

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Neural response telemetry (NRT) measurements are a proven tool for the fitting of cochlear implant (CI) patients. Especially for young children not capable of giving sufficient feedback to electric stimulation during early fitting stages, objective methods as NRT are highly useful for programming the speech processor. Since more and more children are implanted with an auditory brainstem implant (ABI), it is of high interest to apply this technique to the ABI.

Interpretation of NRT signals is technically difficult because the latencies of the responses are shorter than in CI patients. Thus, the first negative peak of the response (N1) appears before the onset of the classic NRT measurement window and cannot be used for the analysis. We simulated the use of only the first positive peak (P1) for the interpretation of NRT measurements in CI patients and found this to be as meaningful as the classic N1-P1 interpretation. The method was then systematically varied in adult ABI patients. In comparison to the linear CI the ABI is a 2-dimensional surface electrode array and each electrode has up to six directly neighboring electrodes for recording an NRT signal. If successively all electrodes are used to record, a two-dimensional picture of the spread of excitation can be obtained. Since electrode migration is a problem shortly after ABI implantation, NRT measurements can also be used as a reproducible “finger-print” to monitor the position of the electrode.

As in a previous study [1] a distinction between auditory and non-auditory electrodes on the ABI cannot be made sufficiently accurate, however, an approximation of the stimulation levels for the ABI programming has shown to be possible. Stimulation levels obtained from NRT measurements were also well accepted by ABI children. A more detailed validation of NRT responses in ABI patients still remains as topic for further studies as besides auditory sensitive neurons also other neurons will respond to electrical stimulation. At the moment this can be done only with well-established methods like electrically evoked auditory brainstem responses (eABR) or patient feedback.

[1] Otto, S.R., Waring, M.D., Kuchta, J. Neural response telemetry and auditory/nonauditory sensations in 15 recipients of auditory brainstem implants Journal of the American Academy of Audiology Volume 16, Issue 4, 2005, 219-227

T13: MIDBRAIN ACTIVITY EVOKED BY PULSED LASER LIGHT

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The possibility of an optical stimulation of neural structures has recently become a research focus in auditory science. Different potential mechanisms for neural stimulation by light exist. In optogenetic approaches neural structures are manipulated to express light sensitive ion channels, while direct optical stimulation with infrared light relies on indirect activation of neurons by a mechanism based on fast, local changes in temperature or pressure. When using laser pulses for neural stimulation in the auditory system, optoacoustic signals produced by the laser in the irradiated media are relevant. The optoacoustic effects themselves could serve as a mechanism allowing frequency-specific movements at the basilar membrane, by that replacing the cochlear amplifier in hearing-impaired ears.

In the present study we compare the frequency pattern of laser induced sounds with the central nucleus of the inferior colliculus (ICc) responses to laser stimulation of the intact and deafened guinea pig auditory system for different wavelengths, pulse energies and pulse durations in vivo. The stimulation was performed through a cochleostomy in the basal turn of the cochlea.

Inferior collicular response patterns to laser pulses in normal-hearing, anesthetized guinea pigs were recorded using multielectrode arrays (32 channels) positioned along the tonotopic axis. Nanosecond laser pulses in the visible and near infrared (NIR) wavelength range (420-2150 nm) at different pulse energies (<10 µJ) were used for stimulation. At 1860 nm the pulse duration was varied from nanoseconds to microseconds. The responses in the ICc were compared to responses to pure tones and 50 µs clicks. Furthermore, the animals were deafened to distinguish optoacoustic responses from direct neural stimulation.

Optoacoustic stimulation results in a heterogenous activation of the iso-frequency laminae of the IC. The evoked firing rate in the midbrain increased with increasing pulse energy. The response pattern shows similarities to that evoked by a 50 µs click for NIR wavelengths. The strength of the response and its dominant spectral components were dependent on absorption coefficient of water, whereas higher absorption resulted in larger spread of excitation in the ICc. Wavelengths with low water absorption elicited no response in the ICc.

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T14: FREQUENCY FOLLOWING RESPONSES TO VOCODED SPEECH

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Vocoded speech offers an innovative platform to study the mechanisms of temporal and spectral processing in the human auditory system. While generating vocoded signals, the low-pass filter cutoff frequency determines the amount of temporal envelope cues while the number of frequency channels determines the spectral cues available. Many perceptual studies (Shannon et al., 1995; Fu et al., 1998; Xu et al., 2002) and a cortical electrophysiological study (Friesen et al., 2009) have examined the effect of varying the temporal envelope cutoff frequency and the number of frequency channels in the signal. Our experiment aimed to extend the existing studies to the level of the brainstem.

We measured subcortical pitch encoding using the frequency following response (FFR) to vocoded stimuli in normal hearing listeners. The scalp-recorded FFR has been shown to reflect brainstem-level neural phase-locking to numerous stimuli such as steady-state vowels (Krishnan, 1999) and time-varying complex speech sounds such as Mandarin tones (Krishan and Gandour, 2009).The questions of interest addressed by this study were as follows: 1) Is the FFR a feasible measure of the subcortical pitch encoding of vocoded speech? 2) Is the FFR sensitive to changes in temporal cues in the vocoded signal? 3) Is the FFR sensitive to changes in spectral cues in the vocoded signal?

In order to answer these questions, FFRs were collected from normal hearing adults listening to an unprocessed vowel /u/ as well as vocoded versions of the vowel with varying temporal envelope cutoff frequencies (50 and 500 Hz with 8 channels) and varying channel numbers (1, 2, 4, 8, and 16 channels with 500 Hz envelope). FFRs were collected in both condensation and rarefaction polarities; addition of the two polarities yielded the FFR to the stimulus envelope while subtraction yielded the FFR to the stimulus fine structure (Aiken and Picton, 2008). Our results showed that the FFR is indeed a feasible measure of the neural encoding of vocoded speech at the brainstem level. The FFR encoding to the stimulus envelope was significantly stronger with 500-Hz than with 50-Hz temporal envelopes in the vocoded signal. Also, the sine-wave vocoder produced a more robust FFR than the noise-band vocoder. As the channel number was increased from 1 to 4, there was an improvement in the FFR encoding of F0, which plateaued from 4-16 channels. These FFR data are consistent with previous perceptual results (Fu et al., 1998; Xu et al., 2002; Gonzalez and Oliver, 2005).

In our knowledge, this is the first experiment looking at subcortical encoding of vocoded speech using the FFR. Understanding the relationship between perceptual and physiological behaviors has a key role to play in developing speech processing strategies for auditory prostheses such as cochlear implants. Characterizing the electrophysiological responses to processed stimuli is also important in order to better understand neural processing of degraded acoustic signals taking place in the normal and the impaired auditory systems. Study of electrophysiological behavior to vocoded speech may facilitate the development of objective clinical tools and tests to evaluate performance of cochlear implants and other auditory prostheses.

T15: ENVELOPE SHAPE AFFECTS NEURAL ITD CODING WITH BILATERAL COCHLEAR IMPLANTS

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Bilateral cochlear implant (CI) users receive minimal benefit from interaural time difference (ITD) cues, especially with present generation sound processors, which deliver ITDs only in the amplitude envelopes of high-rate pulse trains. There are minimal data on the neural processing of envelope ITD cues using electrical stimulation. Therefore, we measured ITD sensitivity to amplitude-modulated electric pulse trains for single neurons in the inferior colliculus of anesthetized, acutely-deafened, bilaterally-implanted cats. The only previous study of neural envelope ITD coding with CI (Smith & Delgutte J Neurophysiol 99:2390) used sinusoidal modulation, which confounds repetition rate and the width of each envelope cycle. Here, we used stimuli that allow independent manipulation of the two variables, and thereby better represent the diversity of envelope shapes contained in natural stimuli.

Specifically, the stimuli were amplitude modulated, high rate (1000 pulses/s) pulse trains. The envelope waveform interleaved single cycles of a sinusoid with silent intervals. Both the duration of each sinusoidal modulation burst (which determines the attack time), and the repetition rate of the modulation bursts was systematically varied (range: from 16 to 250 Hz for rate, and from 4 to 48 ms for width). Each modulation waveform was presented with at least three different ITDs (0, ±400 μs) to assess neural ITD sensitivity. Typically, the ITD was applied to the whole waveform.

Neural ITD coding was relatively poor using continuous sinusoidal modulation, but was generally enhanced by the insertion of silent intervals between modulation bursts. There was considerable variability among neurons with respect to the envelope parameters that maximized ITD sensitivity. In some neurons, ITD coding was best for low repetition rates (long silent intervals) with minimal dependence on envelope width. In other neurons, coding was best for brief envelope bursts with lesser sensitivity to repetition rate. Across our neuron sample, however, the main trends were consistent with human psychophysics (Laback et al. JASA 130:1515, Noel & Eddington JASA in press, van Hoesel et al. JARO 10:557): ITD coding generally improved with longer silent intervals and briefer envelope bursts (i.e. increased attack slopes), and was best for repetition rates in the range of 50-125 Hz. The variability in the neural data suggests that, for any particular envelope shape, information for ITD perception is primarily coded by a subset of the available neurons.

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T16: APPLICATION OF NEURAL NETWORKS IN PREDICTING SUCCESS WITH COCHLEAR IMPLANTS

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Background: With ongoing technological progress and increasing clinical experience the indication for cochlear implants (CI) was extended from complete deafness to severe hearing impairment. For defining the boundary between CI and hearing aids it is increasingly important to predict the success with CI before surgery. In this investigation, the performance of subjects implanted with a CI is compared to their performance prior to the implantation. Based on these data a statistical prediction model can be developed.

Methods: The additional measurements introduced into our clinical routine [1] are done with adults suffering from profound hearing loss or deafness. Prior to surgery the subjects were tested with an optimally fitted hearing aid simulated with the Master Hearing Aid system [2]. Performed speech tests were Freiburg Monosyllables and Oldenburg Sentence Test (OLSA). To assess linguistic skills the visually presented Text Reception Threshold (TRT) Test was accomplished. General state of health, socio-economic status (SES) and subjective hearing were assessed with questionnaires. Six month after surgery, the speech tests were repeated with the CI. Based on these data a statistical model for predicting CI success using a neural network was developed.

Results: So far pre-surgically obtained data sets of 180 CI candidates are available with 148 of them having passed the six month appointment. With the neural network the speech reception threshold in the OLSA was predicted into a framing corridor of ± 3 dB SNR with an accuracy of 75 %.

Conclusion: Between TRT Test, SES as well as - if available - speech perception provided with MHA and later speech perception with CI high correlations were detected, with which the speech perception with CI can be predicted for the first time as score. By using neural networks a prediction of CI success whose accuracy is clinically acceptable seems to be possible.

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[2] Grimm G., Herzke T., Berg D., Hohmann V. (2006): The Master Hearing Aid: A PC-based platform for algorithm development and evaluation. ActaAcustica united with Acustica 92(4), 618-628.

T17: ANIMAL MODELS FOR COCHLEAR IMPLANTATION TO ASSESS COCHLEAR TRAUMA

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Animal models offer a means to assess the effects of cochlear implantation at the cellular and molecular levels as well as at a functional level. Recent advances in electro-acoustic stimulation (EAS) with hybrid implants extend traditional implantation candidacy criteria to patients with functional hearing in the low frequencies. Such candidates benefit from electrical stimulation to the basal cochlear turns encoding the high frequencies, whilst acoustically amplifying the low frequency sounds at the apical turns. The benefits gained through EAS have been somewhat hampered by clinical evidence that a proportion of patients subsequently lose their residual hearing in the implanted ear. The aetiology of this loss is currently unknown. One mechanism thought to be involved is a fibrotic response triggered post-implantation.

To date we have established an animal model of residual hearing in the guinea-pig for implantation via the round window. This model, using gentamicin to ablate hair cells in the basal turns whilst retaining hair cells in the cochlear apex, allows implantation of 8 electrodes using a 14-electrode array. Auditory brainstem responses were recorded prior to implantation and post-implantation to assess function at selected time-points until sacrifice. Two days post-implantation a 10dB shift was seen at low frequencies, which increased to 20dB by one week post-implantation but no further shift was seen at one month post-implantation. Tissue was prepared for histological analysis (immunohistochemistry and electron microscopy). Immunofluorescence analysis with an antibody to CD45, a common leukocyte antigen, revealed extensive positive labelling of cells in the scala tympani of implanted cochleae one week post-implantation suggesting the presence of an inflammatory response and an up-regulation of macrophages. Minimal CD45 immunoreactivity was detected in the contralateral control cochlea.

Additionally, we have developed the necessary surgical techniques to implant mice (C57Bl6, a strain modelling early onset hearing loss). Survival rates have been good (90%) and initial analysis suggests there is fibrotic development around the implanted array as occurs in our guinea pig model.

Both of these models will help identify measures to reduce trauma to the cochlea and improve implant efficacy. Animal models also allow for future research to be carried out using electrode arrays as a means for drug/agent delivery to cochleae to protect residual hearing and enhance patient outcomes.

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T18: SENSITIVITY OF COCHLEAR VOLUME CONDUCTION MODELS TO BOUNDARY CONDITIONS

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Volumetric models of the cochlea have been used to investigate the physiological response to electrical stimulation. However, existing models generally use boundary conditions that do not truly reflect the in vivo situation. Correct boundary conditions are especially important when modeling monopolar stimulation, since the head itself and far-field grounding electrodes are normally not included in the models.

A detailed guinea pig cochlea model was created to study sensitivities in boundary conditions. Electrodes were placed in the first 1.5 turns of the cochlea, with a constant current source applied at either basal, middle or apical locations. Current density in Rosenthal's canal was then measured for each of the following boundary conditions: (1) ground auditory nerve only, (2) ground all outer surfaces, (3) non-uniform, non-zero voltages (based on results from a whole-head model) on all outer surfaces, (4) ground all outer surfaces on a larger surrounding cube, and (5) ground all outer surfaces on a larger surrounding sphere.

Grounding only the auditory nerve produced results which were most different from the other conditions. Since all other boundaries were insulated, injected current was funneled into the nerve. This is not representative of existing monopolar stimulation techniques with far-field grounds, and suggests that models using only the auditory nerve as their ground may be subject to large errors. The other boundary conditions exhibited more comparable patterns along Rosenthal's canal. Cross-turn effects were observed as per other three-dimensional models. There was some spread in the calculated values, with small percentage differences in peak current density at the stimulation site. Variation between conditions increased with distance from the stimulating electrode, with the non-uniform boundary condition reporting more than double the current density towards the apical end when compared with simply grounding all surfaces. These results indicate that uniform boundary conditions may not be sufficient for high-fidelity models. On the other hand, it was also found that larger boundary sizes more accurately recreate the voltage distributions around the cochlea, as seen in the whole-head model. This may reduce inaccuracies in calculated results when realistic boundary conditions are unavailable.

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T19: BIOCOMPATIBILITY OF DEGRADABLE POLYMERS FOR COATING OF THE COCHLEAR IMPLANT ELECTRODE ARRAY

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Drug delivery into the cochlea together with a cochlear implant (CI) is a matter of research already for several years. Several approaches such as fluid-based delivery via a pump, the use of drug reservoirs or coatings have been proposed. Recently, a coating of the silicone surface of CI electrodes with biodegradable polymers (poly(L-lactide) (PLLA) and poly(4-hydroxybutyrate) (P(4HB)) for a dexamethasone delivery has been described (Bohl et al., 2012). These coatings were now investigated for its organ specific in vitro and in vivo biocompatibility and its degradation in the cochlear environment.

A direct-contact test was performed with a fibroblast cell line (NIH/3T3) and freshly isolated spiral ganglion cells (SGC) on both polymers. Furthermore, silicone models of a CI electrode were coated with a thin film of the polymers. In vivo experiments were performed for 1, 2, 3, and 6 months in guinea pigs using these samples to evaluate the effects of the polymer degradation inside the inner ear. Samples were investigated after explantation by scanning electron microscopy in combination with an energy-dispersive X-ray analysis.

No influence of the polymers on cell viability and SGC survival was detected in vitro in comparison to silicone. In vivo, no differences in SGC density between groups implanted with coated and uncoated silicone samples were detected. Only after 6 month a decrease in cell density was detected in comparison to normal hearing (untreated) animals. In all implanted groups a threshold shift was detected by acoustically evoked auditory brainstem response measurements. This threshold shift was larger at 32 kHz than at 8 kHz but no significant differences between the groups were detected.

P(4HB) is characterized by a more gradual degradation which is completed after 6 months whereas PLLA coatings showed some cracks but did not degrade much during this time.
In conclusion, both polymers can safely be used as coatings for CI electrodes but due to the better mechanical stability and optimal degradation behaviourP(4HB) might be the more promising biodegradable polymer for designing a drug delivery system from the surface of CI electrodes.

Support provided by the German Research Foundation SFB/TR37 TP C4.

Bohl A et al., J Mater Sci Mater Med. 2012;23(9):2151-62.

T20: POTENTIAL BENEFIT OF EMULATING THE COCHLEAR TRAVELLING WAVE IN COCHLEAR IMPLANT SPEECH PROCESSORS

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The cochlear travelling wave richly encodes frequency information of an input signal in the cochlear place of maximum excitation (exploited in cochlear implant (CI) speech coding strategies) as well as in temporal aspects of the waveform. Apart from Taft et al, 2009 (Speech Commun., 51, pp. 1114-23), who explicitly encoded travelling wave delays, emulation of travelling waves have not been used in CI speech processors. The present investigation used a modeling approach to consider the potential benefits of cochlear travelling wave emulation in CI speech processing for encoding of pitch.

Space-time neural spike patterns elicited by cochlear electrical stimulation were modeled in three situations: normal hearing (NH), and CI hearing with either a typical channel vocoder (CV) or a travelling wave (TW) speech processor. The NH and TW models had the same first travelling wave stage, but differed in their further stages. The travelling wave stage was an implementation of a time domain hydrodynamic model of basilar membrane (BM) function. The second stage in the NH model translated the predicted travelling wave into neural spike trains using a stochastic model that related BM displacement to neural firing probability. The model produced space-time neural spike patterns from these firing probabilities, given a particular input signal.

The two CI model variations simulated the CI speech processors of which the outputs were space-time patterns of electrical stimulation pulses. CV speech processing implemented an ACE-like strategy that filtered the input signal into frequency bands and extracted signal power within each band to determine the stimulus pulse amplitude. TW speech processing used the travelling wave stage as input, but took CI restrictions regarding temporal and spatial resolution into account. Both these model variations used the same neural model to predict probability of neural firing in reaction to electrical stimulus pulse trains, and could produce space-time neural spiking patterns from an input signal. Inter-spike interval (ISI) histograms were then predicted from the spatio-temporal spike train patterns produced by the three model variations.

For a pure tone input, clusters of ISIs appeared at multiples of the tone period in the NH model. Predictions corresponded to available data. The effect of current spread and resulting loss in spatial resolution was evident in both the CV and TW models. CV speech processing resulted in predicted neural spike trains that were entrained to the electrical stimulus pulse train with ISIs clustering around multiples of the stimulus pulse period (as expected from available data). However, the periodicity of the input pure tone was reflected in spike patterns from TW speech processing, suggesting that input signal periodicity information may potentially be retained. We conclude that travelling wave-like speech processing may have the potential to produce better low frequency information representation in CI-elicited spike trains. This could enhance voice pitch perception.

T21: A NOVEL, AUTOMATED COCHLEAR-IMPLANT INSERTION TECHNIQUE

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Contemporary cochlear implant (CI) insertion technique involves manual insertion of the electrode array into scala tympani (ST), either through the round window or a cochleostomy site antero-inferior to the round window. However, numerous studies have reported that this insertion technique is associated with a high incidence of migration of the electrode array from ST to scala vestibuli (SV), high variability in insertion depths with implant insertions often not deeper than the first basal turn and significant trauma to cochlear structures. While most cochlear implants do achieve their intended outcome of restoring partial hearing to profoundly deaf people, there is a high degree of variability in speech recognition scores across listeners using the same device type. It has been shown that this variability in performance across CI listeners is associated with limitations of current surgical technique that results in variability in insertion depth and location of the electrode arrays. Moreover, high degree of trauma to cochlear structures resulting from manual insertion limits the success of combined electro-acoustic stimulation. Attempts of improving speech and pitch perception with deeper implantation have been met with limited success as manual insertion deeper into the cochlea often led to more insertion trauma to cochlear structures. In the present study we focused on a novel cochlear implantation technique to (1) standardize insertion technique across subjects, (2) achieve deeper implant insertion while (3) minimizing trauma to cochlear structures with a relatively gentle implant insertion.

The technique relies on a viscous fluid to flow the implant into cochlea using ‘reverse perfusion.’ 0.5% Sodium Hyaluronate (diluted with artificial perilymph) was perfused through the cochlea from the round window to a small cochleostomy in SV. Implants were flowed into the cochlea with the perfusing fluid. The method has been developed and recently implemented with a custom-made device, which allows the technique to be fully automated and hence standardized across all cochlear insertions. In initial trials with the device, we have achieved uniform deep implantations with minimal-to-none trauma to cochlear structures. The device is being tested in gerbil in-vivo and endocochlear potential, cochlear microphonic and compound action potentials are typically measured to assess the physiological state of the cochlea during the implantation process. The method has been tested in one human cochlea. While there are significant hurdles to be overcome, the method is promising for its essential simplicity and demonstrated capability for deep, non-traumatic insertions.

T22: AN ECAP DETECTION ALGORITHM BASED ON PRIOR SIGNAL KNOWLEDGE, SUITABLE FOR EMBEDDED COCHLEAR IMPLANT SIGNAL PROCESSING

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Electrically evoked compound action potentials (eCAPs) are routinely used in clinical audiology to set the threshold level of a patient's cochlear implant processor. But the estimation of threshold from evoked potentials is often left to the judgment of experts, which can be time consuming. To circumvent this problem, Neurelec's next-generation cochlear implants will embed advanced real-time computation capabilities to perform signal processing on eCAP measurement stream before returning estimates to the fitting application by telemetry.

In this study, we demonstrate that we can leverage prior knowledge on eCAP latency, amplitude and shape to produce robust estimates of the patient thresholds. An eCAP detection algorithm using a threshold criterion based on a likelihood ratio test has been designed. We assess the validity of the strategy in different conditions. First, using a simulation of a recording with signal and noise, we demonstrate that the likelihood ratio test is equivalent to the F-statistic in the case of a one-sample signal. We then demonstrate that prior knowledge of the signal shape can be used to decrease the minimal SNR for detection. We then investigate using both simulation and analytical results the effect of filtering on the recordings and show how to derive optimal filter parameters for a signal-filter pair.

Finally, we demonstrate these results on actual eCAP recorded in animal, using cochlear implanted guinea pigs and a dedicated stimulation board.

Work supported by Neurelec Corporation

T23: STIMULATION OF THE GUINEA PIG COCHLEA WITH TRAINS OF AMPLITUDE MODULATED LASER PULSES

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Cochlear spiral ganglion neurons can be stimulated with infrared radiation. While the stimulation is spatially selective and can be achieved over weeks without notable damages of the cochlea, it has to be demonstrated that infrared neural stimulation (INS) has the dynamics to encode the temporal structure of speech. In the present study, we further characterize the patterns of neuronal activity to irradiation of the cochlea with infrared pulses.

Single Tungsten electrodes and multichannel electrodes were placed in the central nucleus of the inferior colliculus (ICC) to recorded neural activity in response to INS of spiral ganglion neurons, both in normal hearing and deaf guinea pigs. Aculight diode infrared lasers (wavelength: 1844-1878nm) were used generate the optical pulses. The pulse length was between 50 and 100µs, the pulse repetition rate was between 200 and 500Hz, the modulation frequency was between 10 and 250Hz, and the radiant energy was 0 to ~126µJ/pulse. A 200µm optical fiber was inserted into the cochlea and was coupled to the laser to deliver the radiant energy to the target neurons.

The selected units responded to sinusoidal amplitude modulated (AM) tones, AM trains of clicks, and AM trains of laser pulses with a modulation of their spike discharge. At modulation frequencies of 20 Hz, some units tended to respond with 40 Hz to the acoustic stimuli, but only at 20 Hz for the trains of laser pulses. For all modes of stimulation the responses revealed a dominant response to the first cycle of the modulation, with decreasing number of action potential during successive cycles. While amplitude modulated tone bursts and amplitude modulated trains of acoustic clicks showed similar patterns, the response to trains of laser pulses was different.

From the results of the limited number of units available it can be concluded that timing of the responses is more accurate for acoustical stimulation and that modulation is represented better for acoustical stimulation. Still, it can be shown that in general amplitude modulated INS stimuli are able to produce similar response patterns. Further experiments are required to further quantify the trends seen in the initial study.

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T24: DOES NEURAL SHUNTING LIMIT TEMPORAL CODING BY THE AUDITORY NERVE AT HIGH PULSE RATES?

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Implantees' abilities to detect modulation is typically poorer at higher pulse rates (>2 kHz) than at lower pulse rates [e.g., 1]. The compound neural response recorded in human implantees by Wilson et al. [2] suggests to us that this poor modulation sensitivity may reflect a failure of many neurons to respond to pulses subsequent to the first, consistent with the "neural shunting mechanism," wherein action potential recovery is delayed by the sub-threshold responses or "failed action potentials" induced by these subsequent pulses [e.g., 3]. In this work, we use the mathematical description of spike generation developed by Rothman and Manis [4] to 1) test the idea that neural shunting quantitatively accounts for the temporal response patterns measured in humans at high rates 2) investigate the robustness of this shunting response to the pulse rate and pulse amplitude, and 3) further elucidate the biophysical and dynamical factors that produce this firing blockage.

Because we hypothesize the response patterns measured by Wilson et al. are dominated by excitation and refractoriness, but only minimally influenced by dendritic filtering, we use a single-compartment representation for each neuron in the simulated fiber population. The dynamical equations governing spike generation are those derived from primary-like cells of the cochlear nucleus [4]. This representation was chosen because an equivalently detailed model of spike generation for the auditory nerve is not yet available. As in Wilson et al., the RMS response amplitude to each pulse is extracted via a subtraction procedure and then expressed as a fraction of the initial response amplitude. Preliminary results indicate that the model responses, derived in this manner, agree well with the compound neural response measured by Wilson et al. Our results also indicate that neural shunting is determined by a nonlinear map [5], which quantifies pulse-induced recovery delay, and not by a point process tied to the timing of prior spikes [6].

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T25: WHY SO MANY NEURONS ARE NOT RESPONSIVE TO CI STIMULATION IN AUDITORY CORTEX?

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Understanding how the central auditory system processes sound via a cochlear implant (CI) can be instrumental in guiding new CI technology advances. To study the brain mechanisms involved in acoustic and electric hearing, we have developed a unique non-human primate CI model, the common marmoset (*Callithrixjacchus)*. A multi-channel electrode was chronically implanted in one cochlea; the other remained acoustically intact. This enabled us to record and compare on a neuron-by-neuron basis responses to acoustic, CI or CI/acoustic binaural combination stimuli in the awake condition. After recording extensively in auditory cortex of three implanted animals, we have made an interesting and surprising observation: while most neurons respond to acoustic stimuli, many do not respond to CI stimulation, regardless whether it is delivered from contralateral or ipsilateral ear.

We found that CI-nonresponsive neurons have narrower tuning curves compared to CI-responsive units. Of those CI-nonresponsive neurons, a subset was tested with binaural CI/acoustic stimuli and some of these neurons showed CI-evoked suppression. The CI-suppressed neurons had even narrower tuning than neurons with no CI/acoustic interaction. This might be explained by broad cochlear excitation areas caused by electric stimulation via a CI device, which is ineffective at activating the many cortical neurons with narrow tuning and sideband inhibition. To further assess neurons' responsiveness to narrow and broadband stimuli, bandpass noises, centered at the BF, ranging between 0-2 octaves were presented. While most neurons preferred narrow band stimuli, CI-nonresponsive neurons responded more poorly to broadband stimuli than CI-responsive neurons. This observation further confirms the notion that the lack of responsiveness to CI of many auditory cortex neurons results from broadband nature of electrical stimulation on the cochlea.

Furthermore, we found that there was a greater extent of non-monotonicity in BF-tone evoked rate-level function in CI-suppressed neurons compared to CI-driven neurons or CI-nonresponsive neurons that showed no CI/acoustic interaction. Together, these results show that auditory cortex neurons that are narrowly tuned in frequency or sound level are poorly activated or even suppressed by CI stimulation. Such neurons may be important in fine frequency and level discrimination, and these results might help explain poor CI user performance in such perceptual tasks.

In order to further explore the characteristics of the CI-nonresponsive neurons, we have studied these neurons using complex electric stimuli and developed a two-electrode recording and adaptive filter technique to minimize the stimulus artifacts generated by non-periodic electric stimuli. Using jittered electric pulse trains, we were able to show that some CI-nonresponsive neurons that were non-responsive to periodic electric pulse train with fixed amplitude were found to be responsive to jittered pulse trains. Such insights help us better understand limitations of CI stimulation and develop strategies to overcome such limitations.

Our findings suggest that selective populations of auditory cortex neurons are not effectively activated by CI stimulation because of their complex stimulus selectivity and provide important insights into mechanisms underlying poor performance in a wide range of perceptual tasks by CI users.

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T26: PREDICTING LOUDNESS GROWTH FROM THRESHOLDS MEASURED WITH BROAD AND FOCUSED ELECTRODE CONFIGURATIONS

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Previous modeling efforts in our laboratory have examined the putative quality of the electrode-neuron interfaces across the electrode array of individual cochlear implant listeners. Channels with a high threshold when measured with a focused electrode configuration like tripolar or partial-tripolar were hypothesized to have a relatively poor electrode-neuron interface. The computational model, combining volumetric current spread and activation of auditory nerve fibers across the cochlea, suggested that such channels were affected by poor localized neural survival or a large electrode-neuron distance.

In this study, we describe how a modified version of the model can be applied to predict the growth of loudness on individual channels, with the free parameters of the model (electrode radial position, pattern of viable neurons, global activation threshold) fit only to thresholds measured with the monopolar and tripolar configurations. We found that the prediction of loudness growth slopes could be improved if each modeled neuron is able to contribute multiple spikes over a narrow dynamic range, rather than the simple on/off activation of the previous model. In general, the model was found to corroborate previous psychophysical findings in our laboratory, that channels with high tripolar thresholds exhibit steeper loudness growth functions, whether obtained with broad or focused configurations.

T27: PRIMATE STUDIES OF VESTIBULAR IMPLANTATION: SINGLE NEURON, EVOKED POTENTIAL AND EYE MOVEMENT RECORDINGS

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Introduction: Current vestibular prostheses have been evaluated primarily with eye movement and neural recording in animal models. Although such studies show the promise of this technology, there are many complexities in the response to electrical stimulation that may be underappreciated.

Methods: We implanted 11 monkeys with a vestibular prosthesis designed for human use. We recorded eye movements and neural responses (vestibular evoked compound action potentials -vECAPs, and single neuron responses) longitudinally in response to electrical stimulation alone or during en-bloc canal plane rotation. Eye movements were recorded with scleral coil.

Results: First, there are significant dynamics to the vestibulo-ocular response to electrical stimulation, such that high frequency sinusoidal amplitude modulated stimulation produces higher response gains than low frequency stimulation. Transient stimuli also produce far higher slow phase velocities than longer (>2s) stimulation trains. These response characteristics are reflected in the response properties of vestibular neurons during electrical stimulation, which show doublets and 1:1 phase locking for transient stimuli, but drop out of driven spike discharge for longer stimulus durations. Second, longitudinal data suggest that over many months of stimulation, the gains of the eye movement and neural response to electrical stimulation change significantly, such that there is either an increase or a decrease in the slope of the relationship between slow phase eye velocity and stimulation current or pulse frequency. The elimination of the hair cell input in implanted animals changes the relationship between the evoked neural responses, as measured by vECAP, and the observed slow phase eye movements; i.e., vECAP responses decrease while slow phase eye velocities increase for the same stimulation current. This implies that adaptive gain increases, and not sustained driving of vestibular afferents, are responsible for the long-term efficacy of electrical stimulation. Third, the transfer functions for stimulation current and pulse frequency versus slow phase eye velocity are highly non-linear, and vary over several orders of magnitude between individual animals, or individual canals within an animal. This suggests that optimal mapping of such devices must incorporate a feedforward model of the empirically defined transfer function for each stimulation site, and may need to be remapped over time.

Conclusions: Vestibular prosthesis research has advanced to the point where viable designs exist and can be incorporated into devices for human implantation. However, several observations suggest a level of complexity in the response to electrical stimulation of the vestibular end organ that may need to be addressed for successful long-term implementation of a human vestibular prosthesis.

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T28: TRACKING BRAIN PLASTICITY WITH EVENT-RELATED OPTICAL SIGNAL (EROS)

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The recovery of hearing ability following cochlear implantation provides a unique window for examining the plasticity of the adult human brain. Evoked potentials provide real time information, but without good localization, while fMRI is usually not possible post-implant. In the current study, we tracked the change in a fronto-temporal network in cochlear implant patients using a new modality, the Event-related Optical Signal (EROS). EROS combines spatial resolution of better than a centimeter with temporal resolution of the order of milliseconds, which makes it ideal to study the time course of neural activity in localized cortical areas. The imaging signal reflects changes in the optical scattering properties of brain tissue that are concurrent with neuronal activity.

Five cochlear implant patients with hearing loss onset after 45 years of age participated in three brain imaging sessions, i.e., pre-operative, 2-weeks post-implantation, and 6-months post-implantation. EROS and event-related brain potentials (ERP) were recorded simultaneously in each imaging session. The experiment in each imaging session included three types of blocks: high and low frequency auditory and visual (control). In the auditory blocks, 80% of the tones were 100ms long while the remaining 20% were 50ms long. Visual blocks consisted of vertical (80%) or horizontal (20%) black and white color bars. The task of the participants was to respond to the length of the tones or the orientation of the bars by button press. Larger improvement in the behavioral and brain responses was expected in the high-frequency than the low-frequency auditory block in the post-operation sessions compared to the pre-operation session, while the visual block served as the control for training effects across imaging sessions. Behavioral results showed greater improvement in accuracy for the high-frequency than the low-frequency blocks, which corresponded with an increase in the ERP P300 amplitudes across the imaging sessions. EROS results showed increase in temporal cortex activities in most participants and increase in frontal cortex activities for some participants in the high-frequency blocks. This is the first study to apply EROS to monitor the reorganization of brain responses associated with recovery of hearing ability. It demonstrated the possibility of EROS in brain plasticity research in cochlear implant users and in clinical use.

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T29: LONG DURATIONS OF UNILATERAL COCHLEAR IMPLANT USE DO NOT PROTECT THE UN-STIMULATED PATHWAYS FROM EFFECTS OF AUDITORY DEPRIVATION

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Cochlear implants best restore hearing to children who are deaf when they are provided within limited durations of deafness. It is well known that long periods of auditory deprivation in early life can cause reorganization in the auditory pathways and allow cross-modal reorganization of auditory sensory cortices by non-auditory systems (i.e., visual system, somatosensory system). These changes to the auditory system may be impossible to reverse once a sensitive period in development has passed. In the present study, we asked whether a period of unilateral cochlear implant use protects the auditory pathways from the un-implanted and deprived ear from such effects of deafness.

We tracked the development of the electrically evoked cortical responses from initial activation of the device up to one year after cochlear implantation in 10 children who experienced long durations of bilateral deafness (8.95 ± 4.05 years) prior to implantation. Responses were compared to a group of unilaterally implanted children (n=24) who had similar durations of deprivation on the un-implanted side (9.14 ± 3.05 years). Cortical responses were recorded from 64-cephalic electrodes on the head, using 250Hz biphasic pulse trains presented at a rate of 1Hz and delivered to a single electrode (#20) at the apical end of the implanted array. Cortical generators underlying the evoked peaks were explored using our validated beamformer imaging method, specifically designed to localize areas of cortical activity in cochlear implant users (Wong & Gordon, IEEE Trans Biomed Eng, 2009; Gordon et al., Brain, 2013).

Cortical responses recorded from the deprived ears were similar in both groups and were characterized by an abnormally large negative amplitude peak, labeled N(ci), followed by a large positive peak, P(ci). Underlying activity was widespread in both groups and was of increased intensity relative to the unilateral group's experienced ear. Little change in evoked responses from deprived ears occurred with implant use in either group. These findings suggest that long-term unilateral cochlear implant use does little to protect the un-stimulated cortical pathways from effects of auditory deprivation. The abnormal increase in cortical activity in deprived ears could reflect either: 1) persistent immaturity of the auditory cortex, or 2) reorganization of the deprived cortical pathways.

T30: EFFECTS OF DEAFNESS, CHRONIC INTRACOCHLEAR ELECTRIC STIMULATION AND BEHAVIORAL TRAINING ON NEURONAL TEMPORAL PROCESSING IN ICC AND AI

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Deaf humans using a cochlear implant (CI) depend largely on temporal cues for speech recognition because spectral information processing is severely impaired. Auditory experience and training can enhance electric hearing performance even in prelingually deaf, adult subjects, suggesting that 1) relevant auditory experience modifies central auditory temporal processing and 2) functional plasticity is not limited to a critical period during development. This study compares the effects of intracochlear electric stimulation (ICES) and behavioral training on temporal processing in the central nucleus of the inferior colliculus (ICC) and the primary auditory cortex (AI) in cats with different ages at deafness onset and deafness durations. Kittens were deafened neonatally by injection of an ototoxic drug and implanted unilaterally with a feline prosthesis, either as juveniles or as adults after long-term deafness (>3.5 yr). Passive ICES was initiated in all experimental groups, and some animals also received behavioral training on a signal detection task. Adult deafened animals and long-deafened unstimulated animals served as controls. Neuronal responses in the ICC and the AI to unmodulated pulse trains were recorded with tungsten microelectrodes in barbiturate anesthetized animals. The temporal response parameters of interest were best repetition rate (BRR) that produced the maximum number of phase-locked spikes, the cutoff rate at which the number of phase-locked spikes was just less than 50% of that at BRR, the entrainment of spikes to single pulses, and the minimum neuronal response latency and jitter.

Our analyses showed that long-term deprivation of auditory input resulted in severely reduced temporal resolution (lower repetition rate following, lower entrainment, longer latencies) and response precision (increased jitter) in both ICC and AI. However, passive ICES improved degraded temporal processing in long-deaf cats in both auditory structures. When compared to passive stimulation alone, behaviorally relevant ICES did not affect temporal processing in the ICC but significantly enhanced temporal resolution and response precision in AI of both juvenile and long-deaf cats.

Our findings indicate that behavioral training has a powerful effect on profoundly degraded cortical temporal processing in the deaf auditory system. This potential for functional (temporal) plasticity is not confined to critical periods during development. The described improvements in cortical temporal processing may be a physiological correlate for perceptual training-induced improvements in speech perception and language-processing abilities in human CI users.

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T31: PITCH PERCEPTION IN COCHLEAR IMPLANT USERS WITH SQUARE WAVE ITERATED RIPPLED NOISE

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Manipulation of temporal (rate) pitch mechanisms in cochlear implant (CI) users through carrier rate modulation may expand the potential number of distinct pitch percepts available beyond that which is available through place pitch coding alone. Rate pitch has the advantage of being able to provide a continuum of pitches on a single electrode up to approximately 300 Hz. However, this limit varies greatly between subjects and within subjects (for different electrodes), with a few exceptional performers being able to distinguish rate pitch cues up to at least 1000 Hz. Square wave iterated rippled noise (SWIRN) was created through direct electrical stimulation.

CI subjects were asked to identify the higher of two tones generated with direct electrical stimulation using SWIRN . The base rates ranged from 100 Hz to 500 Hz; target rates were 35% higher. The carrier stimulation rate was 15 kHz. Using a method of constant stimuli, pitch ranking was measured for three base modulation rates on groups of 11, 6 and 3 electrodes on a Cochlear Ltd. device. These groups consisted of one, two or three basal, middle and apical electrodes. The order of the electrodes was randomized in each block. In SWIRN, the current level was ranged from 0%-100% of the dynamic range. To compare performance of SWIRN pitch ranking was also measured with pulse frequency modulation (PFM) on one electrode (single pulses per period SPP) or on multiple electrodes (multiple pulses per period MPP). Preliminary results demonstrate little variability across subjects, frequencies and electrode groups. Increasing the amount of electrodes stimulated does not automatically lead to better performance. The ranking performance was significantly above chancel level on all tested rates when MPP stimulation was applied to 11 electrodes. SPP stimuli had low pass characteristic and SWIRN stimuli did not improve pitch ranking scores no matter how many electrodes were stimulated.

If a sound processing strategy were to use better temporal coding then possibly pitch will be better perceived by implant users, yielding improved language and music perception.

T32: ELECTRO-ACOUSTIC PITCH MATCHING IN BIMODAL COCHLEAR IMPLANT USERS

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Background: Pitch perception is usually poor in cochlear implant users, and some studies have shown that aspects of sounds like intensity and timbre can influence pitch perception in CI users. Recent studies (Green et al. 2011, Carlyon et al. 2010) have shown that the effect of intensity on perception of pitch is greater than already expected. In addition to this possibility, differences in pitch perception between hearing aids and cochlear implants make the perception of pitch more complicated for people using bimodal hearing. These factors may hinder the enjoyment of music. This study aimed to investigate the perception of pitch for sounds with different timbre and intensity in patients with bimodal hearing.

Methods: 12 bimodal CI users participated in this experiment, all fitted with the ACE strategy in the implanted ear. The task was matching of sounds in the non-implanted ear to reference sounds which were always presented to the implanted ear. The reference sounds were presented at different levels of intensity (soft and loud) and with different types of timbre (single electrode, pure tone, and piano note). These reference sounds were chosen to investigate the effects of current spread around a single electrode, and the effects of electrode activation changes in the sound processor as intensity increased for pure tones and for piano notes. The pitch matching was done for low and high frequencies (250 Hz and either 750 or 1000Hz depending on residual hearing thresholds in the non-implanted ear).

Results: Statistical analysis showed that the frequencies of the matched sounds were different for different types of reference sounds, and that this effect was highly significant. Intensity also changed the matched frequency for some users by up to ten semitones. A large inter-subject variability was observed for all types of sounds and the change in matched frequency between soft and loud sounds depended on the type of reference sound.

Conclusion: Timbre and intensity levels of sound can change the perception of pitch for CI users, potentially leading to pitch mismatch between the two ears of bimodal listeners as intensity and timbre vary. These factors should be taken into consideration in the effort to improve music perception for cochlear implant users. This is particularly important because changes in intensity and type of sounds are more common in music than in speech.

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T33: PERCEPTION OF COCHLEAR-IMPLANT SIMULATED SPEECH BETTER IN MUSICIANS

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In normal-hearing (NH) listeners, musical experience has been associated with better speech understanding in noise (Parbery-Clark et al. 2009, Ear Hear. 6, 653-661), presumably because better perception of pitch cues help to segregate speech from noise. Because of the limited spectral resolution, cochlear implant (CI) patients have difficulty understanding speech in noise and perceiving melodic pitch. Musical training may improve CI patients' melodic pitch perception, which in turn may improve speech understanding in noise. However, the degradation of spectral cues associated with CI signal processing may limit the contribution of musical experience to speech performance. In this study, speech understanding in noise was measured in NH musicians and non-musicians while listening to unprocessed stimuli or stimuli processed by an acoustic CI simulation.

Ten musicians and five non-musicians participated in the study. All had normal hearing and the mean age was 22 years for the musicians and 27 for the non-musicians. Musicians were defined as: 1) having begun musical training before age 7, 2) having 10 years or more of musical training, and 3) having received some music training within the last 3 years. Non-musicians were defined as, in addition to not meeting the musician criteria, not having received musical training within the 7 years before the study. Recognition of monosyllable words (NVA test) was measured in quiet and in steady, speech-shaped noise at three signal-to-noise ratios (SNRs): 0, +5 and +10 dB. Word recognition was measured while listening to unprocessed speech or to an 8-channel, sinewave-vocoded CI simulation.

Preliminary data show an advantage for musicians when listening to the CI simulation at the 0 dB and +5 dB SNRs. With unprocessed stimuli or with the CI simulation in quiet or at the +10 dB SNR, there was no difference between musician and non-musician performance. The preliminary data suggest that musicians may be better able to access subtle cues than non-musicians when the system is pressed (i.e., poor spectral resolution combined with poor SNRs). Under favorable listening conditions (good spectral resolution and/or high SNRs), music experience does not seem to contribute to performance. The results suggest that a musical training may not only improve CI patients' music perception, but also improve speech understanding in noise.

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T34: TRAINING TO IMPROVE PITCH PERCEPTION OF COCHLEAR IMPLANT USERS IN THE PRESENCE OF COMPETING PLACE CUES

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Perception of musical pitch coded by modulation rate in present cochlear implants (CI) can be adversely affected by changes in spectral timbre (coded by place of stimulation). In this study we examine whether the influence of place on rate-pitch can be reduced through training. The specific hypotheses are:

1) isolated training on a single cue (i.e. pitch or spectral timbre) in the absence of other cue variations will improve listeners' sensitivity to that cue even in complex sounds in which multiple cues vary; and
2) subsequent training with complex sounds in which multiple cues vary, further improves listeners' ability to discriminate the pitch of complex sounds.

A computer-based training program was developed that required subjects to match acoustic patterns of pitch and/or spectral timbre to visual patterns of notes on a musical staff. Ten adult CI recipients participated in the study: five used the training program and five underwent standard rehabilitation and acted as controls. Recipients allocated to each group were balanced in terms of prior musical experience and modality of hearing.

Prior to training, fundamental frequency (F0) and resonant frequency discrimination for stimuli presented in the sound field were measured in all subjects using their standard clinical device(s). The F0 tests were conducted with and without variations in place information. The trainees took the training program home for a period of four months, over which they were asked to train for approximately 30 minutes/day using their standard clinical device(s). The training program comprised two phases that addressed each hypothesis in turn. Each phase was administered over a two month period. Within each phase the complexity of stimuli and patterns increased as training progressed, and the interval between stimuli (e.g., F0 for the pitch) was adapted based on trainees' performance. Synthetic complex harmonic tones were used to allow for precise control of the acoustic stimulus. In the second phase additional samples of real instruments, including piano, saxophone, trombone and a male and female singer were employed to increase stimulus diversity and relevance to everyday listening.

Discrimination thresholds were remeasured in all subjects at the middle and end of the four month period. Significant improvements in discrimination thresholds were observed for tests conducted midway through the study (after phase 1) for the trainees, but not the controls, which support the first hypothesis. Neither group of subjects showed much improvement over the second training phase. That outcome suggests that either performance of the trainees had saturated, or that training with complex sounds is relatively ineffective compared to training with single cues. Further evaluations of trainees three months after training had ceased showed that some of the improvements were retained suggesting that lasting practical benefits can be obtained.

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T35: MELODIC PITCH PERCEPTION AND LEXICAL TONE PERCEPTION IN MANDARIN-SPEAKING COCHLEAR IMPLANT USERS

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Contemporary cochlear implant (CI) devices provide only limited pitch cues, which are important for music perception and tonal languages such as Mandarin Chinese. Previous studies have reported a strong correlation between Chinese CI users' lexical tone perception and pitch discrimination thresholds (Wang et al., Laryngoscope, 2012.). However, little is known about how music perception, especially melodic pitch perception, relates to Chinese CI users' lexical tone perception. The present study compared melodic contour identification (MCI) and lexical tone perception in young Mandarin-speaking CI users.

MCI and lexical tone perception were measured in 34 young Mandarin-speaking CI users (11 post-lingually deafened and 21 pre-lingually deafened). MCI stimuli were melodic contours composed of 5 notes of equal duration (300 ms); the interval between successive notes in each contour was varied between 1 and 6 semitones. The 9 test contours represented different changes in pitch (rising, falling, flat, rising-falling, falling-rising, flat-rising, flat-falling, rising-flat, falling-flat). Subjects were asked to identify the test contour from among 9 response choices. MCI performance was acutely measured with no training or feedback. Lexical tone recognition was measured for four tonal patterns: tone 1 (flat F0), tone 2 (rising F0), tone 3 (falling-rising F0), and tone 4 (falling F0). Four Mandarin Chinese single-vowels (/a/, /e/, /u/, /i/) were produced by two females and two males with each of the lexical tones. Subjects were asked to identify the test tone from among 4 response choices.

Overall MCI performance was generally poor (23.21% correct). Post-lingually deafened CI subjects performed significantly better (32.32% correct) than pre-lingually deafened CI subjects (18.43% correct). Even with 6-semtone spacing, pre-lingually deafened subjects were able to correctly identify only 19.56% of contours; in contrast, post-lingually deafened subjects were able to correctly identify 41.42% of contours. There was no significant difference in overall lexical tone identification between post-lingually-deafened (81.11% correct) and pre-lingually deafened CI subjects (80.88% correct). MCI and lexical tone identification were significantly correlated in post-lingually deafened CI subjects, but not in pre-lingually deafened CI subjects. The results suggest that Mandarin-speaking Chinese CI users have great difficulty perceiving melodic pitch. Co-varying amplitude and duration cues in the lexical tones most likely compensate for CI users' poor pitch perception. Previous acoustic hearing experience seems to benefit melodic pitch perception by CI users. Music training may improve melodic pitch perception for both pre- and post-lingually deafened CI patients.

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T36: EVALUATION OF MUSIC PRE-PROCESSING SCHEME WITH COCHLEAR IMPLANT RECIPIENTS

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Although cochlear implant (CI) recipients reach good speech understanding in quiet surroundings, music perception and appraisal generally remain poor. Simple musical structures, lyrics that are easy to follow and a clear rhythm/beat were reported amongst the top factors that enhance musical enjoyment for CI recipients [1]. A negative correlation was found for CI recipients between complexity and appraisal of complex songs using pop, country and classical music [2]. Pop and country music was appreciated more than classical music, most likely due to the presence of lyrics and a strong, simple beat. Lyrics and beat are more suitable for transmission through current-day implants than the structural features of instrumental classical music.

The preference for clear vocals and strong rhythm/beat for post-lingually deafened CI recipients was investigated in a previous study, by modifying the relative instrument level settings using multi-track recordings of pop music [3]. A significant difference in preference rating scores was found between normal hearing (NH) and CI subjects. CI subjects preferred an audio mix with larger vocals-to-instruments ratio compared to NH subjects. In addition, given an audio mix with clear vocals and attenuated instruments, CI subjects preferred the bass/drum track to be louder than the other instrument tracks. The relative instrument level settings were modified by altering the levels of the different, separately recorded, instrument tracks, which, however, are usually not available.

In this study, we present a music pre-processing scheme for CI recipients in which vocals and percussion are elucidated. The algorithm is capable of modifying the relative instrument level settings in complex music. Based on the prior work, the hypothesis was that this algorithm would improve the music appreciation of CI recipients. Subjective evaluation of the music pre-processing scheme in the form of blind preference rating will be presented with post-lingually deafened CI recipients.

This work was supported by a Baekeland PhD grant of the Institute for the Promotion of Innovation through Science and Technology in Flanders (IWT090274) and Cochlear Technology Centre Belgium.

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T37: THE EFFECTS OF SOUND CODING PARAMETERS ON THE PERCEPTION OF PROSODY BY COCHLEAR IMPLANT USERS

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Novel sound coding variants for cochlear implants (CI) with enhanced spectral and temporal processing (STEP) were developed with the aim to improve the perception of prosodic information carried by voice pitch. A dual filter-bank approach was employed with a bank of narrow, good quality filters (SP) and a bank of parallel wide filters (TE). SP filters reduced redundancy between channels and allowed for better extraction of the low harmonics of voiced signals compared to the standard ACE strategy. TE filters had increased modulation bandwidth to better represent periodicity in the extracted temporal envelopes for F0 < 375 Hz. Channel selection and non-simultaneous “pitch” steering were provided by SP filters while the amplitudes of regularly timed current pulses 900 pps/ch were modulated across channels using the envelopes obtained via corresponding TE filters.

In a first phase performance was evaluated with 4 Nucleus CI recipients. The same standard Nucleus 22 channel frequency to electrode allocation function was used with STEP and for the comparison ACE strategy with 10 maxima. Voice prosody perception was evaluated with a female speaker (F0 180-500 Hz) using the French Question/Statement task developed by Patel et al. (2008, Music Perception 25(4), 357-368). Further vowel pitch ranking was evaluated for the F0 range 126 to 212 Hz. Subjective sound quality judgments on a scale 0-10 were collected for several speech and music stimuli coded using various strategy parameters.

Results indicated that STEP improved vowel pitch ranking in the range 160 to 180 Hz compared to ACE. However for F0 >180 Hz pitch ranking was not reliable for either strategy. Furthermore when modulation at F0 was removed subjects reported that male voices with low F0 sounded like female voices. There was a slight trend of benefit of using STEP versus ACE for Question/Statement discrimination (d’=1.75 vs. d’=1.00) for the best performing subject. However overall STEP did not provide a high level of discrimination across subjects (mean d’ = 1.1).

Increasing temporal modulation bandwidth, depth and synchrony across channels in the multi-electrode coding of real signals does not appear to improve voice pitch perception for F0 >~180 Hz and does not appear to provide functional benefits (e.g. Milczynski et al, 2009, J AcoustSoc Am 125, 2260-71). Here modulation at F0 >~180 Hz sounded unmodulated using 900 pps/ch carriers, and does not appear to support pitch ranking even with higher rate carriers (1800 pps/ch, Milczynski, 2009).

Therefore in a second phase SP frequency-to-electrode allocation functions were modified to resolve harmonics for F0 >150 Hz up to 1.2 kHz, producing at least one inactive channel between active channels: Can CI subjects use resolved spectral cues to perceive voice pitch? Preliminary data will be available at the meeting.

Study sponsored by Cochlear Europe.

T38: HOW DOES ELECTRIC HEARING COMBINE WITH ACOUSTIC HEARING FOR SPEECH AND MUSIC?

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Adding low frequency acoustic information via a hearing aid (HA) to information from a cochlear implant (CI) has been shown to improve speech performance in noisy backgrounds and familiar melody recognition. Anecdotally, many of these “bimodal” listeners say that the sound quality with both devices combined is much preferable to that with a CI alone, especially for music. In the present study, we tested speech understanding in noise and melodic contour identification (MCI) in three conditions: HA only (acoustic, A), CI only (electric, E), and both together (electro-acoustic stimulation, EAS).

Eight bimodal CI subjects (HA in one ear and CI in the other) were tested using their clinical settings for both devices. Speech reception thresholds (SRTs) were adaptively measured for HINT sentences in a speech babble background. MCI was measured with and without a competing instrument. The 9 target contours were played by a MIDI piano with notes ranging from A3 (220 Hz) to A4 (440 Hz). The competing contour (always flat) was played by a clarinet, which provided a possible timbre cue. The note of the competing contour was A3 (220 Hz; “overlapping”) or A6 (1760 Hz; “non-overlapping”).

Preliminary results showed great inter-subject variability in the integration of EAS. There was a strong difference between speech and music performance, with EAS offering a greater benefit for speech, but little benefit (and even interference) for music (for this particular task). For most subjects, HINT SRTs were poorest with A-only, better with E-only, and best with EAS; however, the difference between E and EAS was not significant. Most subjects had difficulty with the MCI task with E-only (<40% correct), and most had no difficulty (>75%) with A-only. For music, some subjects attended to the better ear (EAS = A); others had difficulty combing information across ears, with some subjects experiencing interference (EAS < A). For most subjects, performance was better when the notes of the masker and target contours did not overlap. As MCI performance was mostly driven by the HA alone, there was little difference in performance between the no masker and the non-overlapping masker conditions, because the masker frequency was beyond the range of the HA.

The results suggest that most speech information is provided by the CI and most low frequency pitch information is provided by the HA. The HA helps with music perception and appreciation and may improve the sound quality. In real world listening conditions, subjects report that the HA provides warmth while the CI provides clarity. Bimodal listeners may require optimized settings for both the CI and HA to reduce interference and maximize sound quality.

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T39: PITCH PERCEPTION AND PRODUCTION FOR HEARING-IMPAIRED CHILDREN WITH DIFFERENT HEARING DEVICES

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Children using a cochlear implant (CI) listen with degraded pitch information which should not only affect pitch perception but also production because of the feedback mechanism for monitoring utterances [Mürbe et al., 2002; J Voice 16, 44-51]. This study looked at whether children with bilateral CIs had different pitch perception and production skills to their normal hearing (NH) and hearing-impaired (HI) peers that used different hearing interventions.

53 children and their parents/caregivers participated in the study. Children were aged between 4 years and 8 years 11 months and the mean age was 6 years 10 months. 13 children had NH, 14 had bilateral cochlear implants (BCI), 8 had a unilateral implant with a contralateral hearing aid (CHA) and 18 used bilateral hearing aids (BHA).

Families completed a questionnaire looking at the development of musical aspects (Musical Stages Profile (MSP)). For perception of pitch the children were assessed with the “Primary Measures of Music Audiation” which contained a “same/different” pitch perception test with complex stimuli and a rhythm task. For production of pitch the children were evaluated with the “National (UK) Child Singing Assessment” in which scores are assigned to: 1. Comfortable singing range, 2. Singing Accuracy, as well as 3. Children’s habitual pitch centre for speech. Laryngograph and audio recordings were made of the utterances. The audio recordings were evaluated by trained listeners to determine the singing accuracy score. Acoustic analyses of the laryngograph recordings were conducted to look at voice quality and speech fundamental frequency measures.

The results for the MSP were analysed using a series of Welch’s F tests (within one-way ANOVA) with intervention group as a factor (NH, BCI, CHA, BHA) and using a significance level of $p<0.05$. Post-hoc comparisons were carried out using the “Least Significant Difference” (LSD) test. The different ANOVAs were conducted for the overall score on the MSP and also for the different domains (sound awareness, emotional aspects, melody and dynamics, rhythm).

There was a significant difference between the NH group and all the HI groups for the overall MSP score and for the domains of melody and dynamics and also for rhythm. There were no significant differences between the HI groups. There were no significant differences for sound awareness or emotional aspects between any groups.

The Welch’s F test with LSD post-hoc showed that there was a significant difference between NH and all HI groups for singing accuracy, there was no significant differences for habitual speech frequency between any of the groups and the only measure where the BCI group was significantly different to the others was for singing range, which was narrower than the other groups.

The statistical analyses of the voice quality results showed that the BCI users had breathier voices than the CHA and NH groups. The results from the PMMA showed that there were no significant differences for rhythm perception but for pitch perception the NH group were significantly better than all HI groups.

In summary the BCI users perform at the same level as their HI peers for many of the pitch perception and production measures used in this study but are poorer with respect to pitch range of singing and they use a breathier voice quality.

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T40: NEW MUSIC COMPOSITIONS FOR COCHLEAR IMPLANT USERS

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In the same manner as in previous concerts [1] [2], we investigated the process of creating and evaluating a music concert of new compositions specially designed for cochlear implant (CI) users. Music should be enjoyed by everybody, but still today, music perception with CI devices is severely limited. In this initiative we try to make a step to improve the integration of cochlear implant users into social-musical activities.

musIC ("música para Implantes Cocleares") is a project where we created new pieces that can be experienced in a similar way by normal hearing listeners as well as by CI users. First we investigated how music is perceived in a group of CI users. Each CI users performed tasks on instrument identification, melodic contour identification and rhythm perception. Additionally, each study participant filled questionnaires about their musical background and they provided us with collections of the music they usually listen. This music was analysed and compared to the music that normal hearing listeners listen using a computational model, the results showed little differences between both groups. In a second phase, a series of three seminars were organized where music composers, CI users and researchers interacted to each other to create 7 music pieces [3]. Finally, in a third phase, the new compositions were interpreted in a concert where the appreciation of the music was evaluated by means of a questionnaire. The music questionnaire was adapted from the one designed by Innes-Brown, 2012, that assessed cognitive aspects of music (interest, emotion and understanding), enjoyment, and technical aspects for each of the new pieces (melody, rhythm and timbre perception).

255 people attended the concert and 133 filled the music appreciation questionnaire (86 normal hearing listeners, 39 CI users and 8 hearing aid users). As expected, the group of CI users rated melody, timbre and rhythm perception significantly lower than the group of normal hearing listeners, however no significant group differences were found on measures of enjoyment and cognitive aspects such as interest, emotion and understanding. This result shows the success of the compositions. For the 11 CI users that participated in the seminars, we investigated the correlation between their music perception abilities, their questionnaire ratings and the results of a model that predicts the complexity of the composed songs.

In general the subjective feedback after the concert was very positive, for some of the cochlear implant users this was the first time they attended a music concert after their cochlear implantation.

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T41: THE BENEFIT OF A UNILATERAL COCHLEAR IMPLANT IN PATIENTS WITH REGULAR HEARING OF THE CONTRALATERAL EAR

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Patients with unilateral deafness are increasingly considering the cochlear implant for their deaf ear for hearing rehabilitation. There are potential benefits from this, as the ability to hear sound from the affected side and listen to voices. The speech perception may improve in difficult listening situations as in noise. The listening may become less strenuous. A persisting tinnitus may become less noticeable. But of course, there are risks of the procedure, the costs and a potential mismatch of the modalities involved.

19 CI recipients (11 with SONATA/CONCERTO + OPUS2, 7 with CI24RE(ST) + CP810 and 1 with HiRes90K + Harmony) were included after informed consent and ethical approval. All were adults and had at least 6 months of CI experience.

We tested the B.I.L.D. (Binaural Intelligibility Level Difference) with noise to the acoustic hearing ear and speech from the front (S0N90). Here, we could observe an improvement of the SRT of 6 dB in the binaural mode. This effect was present in binaural controls, too. In the CI recipients group the individual test results had a broader range.

In the spatial discrimination, the acuity of the detection was 11% with the CI alone, 17% with the hearing ear alone and 27% in the bimodal situation. The vector of the error was significantly decreased with the bimodal situation.

In the Alternate Binaural Loudness Balance testing (A.B.L.B.), the range of equal loud perceptions is between 49 and 55 dB with no statistical significance.

The pitch matching between the ears in CI-recipients as in the controls lies in the average within a musical third (20% frequency difference).

In these 19 patients, we could see measurable benefits from the re-use of the second ear.

T42: AN ALGORITHM TO ENHANCE THE BINAURAL CUES AVAILABLE TO BILATERAL COCHLEAR IMPLANT USERS

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Bilateral cochlear implant (BCI) users have shown reduced binaural abilities as compared to listeners with normal hearing. Although they have been shown to be sensitive to interaural differences of level (ILDs), their sensitivity to interaural time differences (ITDs) is poorer, and as a group they have shown limited ability to improve speech understanding of a target talker in the presence of a spatially-separated distractor talker. A method for enhancing the binaural representation available to BCI users is investigated, which involved computing instantaneous ITDs in the low-frequency region (which are present naturally in a BCI configuration, although not consistently useable by BCI users), and converting them to low-frequency ILDs (which are not present naturally, but are useable, at least by normal-hearing listeners).

The efficacy of such a manipulation on speech intelligibility is evaluated by applying either natural or enhanced binaural cues to both target and distractor talkers to simulate spatial separation. Natural binaural cues were applied by first, using a KEMAR manikin to record a broadband noise off to one side, filtering the recordings into 32 ERB-wide bands, and computing an ILD in each band. This ILD-by-frequency function was then applied to each speech token, thus simulating naturally-occurring ILDs. A whole-waveform ITD was then applied by shifting either the left or right channel by 600 μs. The target talker was always to the left, the distractor talker always to the right. Target and distractor tokens were then mixed, and this mixture represented the stimulus in the Natural condition. For the Enhanced condition, the target-plus-distractor mixture created in the Natural condition was subjected to the ITD-to-ILD conversion algorithm, which estimated the instantaneous ITD in ERB-wide bands below about 2000 Hz, and converted them to instantaneous ILDs, using a simple conversion table in which an ITD of ± 600 μs equaled an ILD of ± 30 dB, and an ITD of 0 μs equaled an ILD of 0 dB. Speech intelligibility was then measured in both conditions.

BCI patients showed substantially greater speech intelligibility with enhanced cues than with naturally-occurring binaural cues. For five BCI patients, average speech intelligibility was 35 percent correct in the Natural condition, and 66 percent correct in the Enhanced conditions. Thus, it is possible to enhance the binaural representation to achieve greater speech intelligibility for BCI users.

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T43: THE BINAURAL INTERACTION COMPONENT OF THE CORTICAL AUDITORY EVOKED POTENTIAL IN CHILDREN

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Bilateral implantation is becoming increasingly common. In order to benefit fully from bilateral implantation, listeners must be able to integrate information from both ears (i.e. binaural processing). Electrophysiological measures such as the binaural interaction component (BIC) of the cortical auditory evoked potentials (CAEP) can be used to evaluate binaural processing. However, little is known about the BIC of the CAEP in children with normal-hearing or cochlear implants (CIs). This study aimed to investigate developmental effects on the BIC of the CAEP in normal-hearing children and in children with CIs.

The BIC of the CAEP was recorded in normal-hearing children and children with bilateral CIs using a multi-channel Neuroscan system. Participants ranged in age between 1.5 and 18 years. In normal hearing children, the stimulus was a narrow-band Gaussian noise presented at 70 dB SPL. In CI children, the stimulus was a 100-ms train of biphasic current pulse delivered to individual electrodes at the C level. The CAEP was recorded in response to left monaural, right monaural and bilateral stimulation for each listener. The BIC of the CAEP was computed by subtracting the CAEP recorded in response to binaural stimulation from the algebraic sum of the two monaural evoked potentials.

Our results showed that the BIC of the CAEP can be recorded from listeners as young as 1.5 years. The BIC of the CAEP demonstrates maturational changes in both children with normal hearing and children with CIs.

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T44: BILATERAL LOUDNESS BALANCING AND DISTORTED SPATIAL MAPS IN RECIPIENTS OF BILATERAL COCHLEAR IMPLANTS

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Use of bilateral cochlear implants (CIs) can provide better sound-localization abilities than monaural CI use. In bilateral CI recipients using clinical speech processors, the primary cue to sound localization is interaural level differences (ILDs). Thus, an ideal procedure for mapping of bilateral CIs would represent ILD cues in a manner that provides a consistent spatial map for sound-source location. Current mapping procedures for bilateral CIs consist predominately of obtaining 'most-comfortable' (C) levels in each ear independently. However, such procedures can sometimes result in distorted spatial maps for ILD cues, in that an ILD of zero can elicit a non-centered image in some bilateral CI recipients [Goupell et al., 2013, J. Acoust. Soc. Am. 133(2) EL101-107]. Here, we investigated whether these distortions could be alleviated through careful within- and between-ear loudness balancing in two related experiments.

In both experiments, 14 bilateral CI users were provided with simultaneous bilateral stimulation. Three pairs of electrodes of the same number were chosen: one in the basal region of the cochlea, one in the middle, and one in the apex. Then, for each electrode in each ear, C-levels were obtained using procedures similar to those used in the clinic. These levels were then carefully loudness balanced within and across ears. The number and location of the perceived intracranial image was then measured for an electrode pair simultaneously stimulated in both ears. ILDs of 0, 5, and 10 current units (CUs) were used to observe the extent to which the auditory image shifted with changes in ILD. In the first experiment, we quantified the number of perceived images and the general location of those images. In the second experiment, we more closely assessed the perceived intracranial location as a function of ILD.

In the first experiment, our results indicate that careful loudness balancing generally produced small changes to the CI maps obtained with conventional fitting procedures. More important, equal loudness ensured a centered intracranial image on only ~50% of trials. On ~20-25% of trials, individuals responded that they heard multiple auditory images, indicating that the signal provided to each ear was not fused. In the second experiment, ~30% of lateralization curves were perceived as significantly offset from center. These data suggest that loudness balancing both within and across ears 1) may not always elicit percepts thought to be optimal for sound location and 2) may yield distorted spatial maps in some individuals. Thus, it may be necessary to develop mapping procedures for bilateral CIs that are spatially as well as loudness based.

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T45: SELF-SELECTED AND PITCH-MATCHED FREQUENCY TABLES IN USERS OF BILATERAL COCHLEAR IMPLANTS

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Two benefits of bilateral cochlear implant (CI) use are improved speech-understanding abilities in noise and sound localization. These benefits may be less likely, however, if a given signal stimulates neural elements in each ear that are tuned to different characteristic frequencies. Such a “bilateral mismatch” might be observed if there is a between-ear difference in electrode insertion depth or neural survival and may manifest itself perceptually as a difference in pitch between ears. Unfortunately, there are few tools that can be used to identify whether an individual has a bilateral mismatch, or to determine how the CI programming can be modified to resolve the issue. To address these issues, we have been exploring the use of real-time or near real-time frequency-table selections as a tool to identify cases of bilateral mismatch. Here we investigated whether frequency-table selections made to maximize bilateral speech intelligibility were the same as those chosen to match the pitch percepts heard in each ear.

All participants were presented with speech stimuli, and adjusted the frequency table until speech intelligibility was perceived as being maximized. These selections were made unilaterally, and while also listening with the contralateral CI. Then, participants were presented with a pure tone. While listening simultaneously with both ears, the frequency table was adjusted in one ear until the pitch percepts were matched. Approximately 8 pure tones were presented for each participant.

When selecting a frequency table thought to maximize speech intelligibility, approximately 50% of bilateral CI recipients selected a table in one ear that differed from the standard table worn in the contralateral ear. Moreover, the tables chosen to match the pitch percepts elicited in each ear were generally shifted higher in frequency than the table chosen to maximize speech understanding. These results suggest that some bilateral CI recipients may have bilateral mismatches to which they have not fully adapted, and that these mismatches can be identified through self-selection of a preferred frequency table. The differences between the speech-selected and pitch-matched tables suggest that when making these selections, listeners are trying to select a signal that is as natural-sounding as possible without hindering speech intelligibility.

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T46: LATERALIZATION AND BINAURAL FUSION FOR LONG INTERAURAL TIME DELAYS (ITDS) WITH BILATERAL COCHLEAR IMPLANTS

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Most bilateral cochlear implant (BiCI) users are sensitive to ITD; however the neural mechanisms underlying this sensitivity are not yet understood. We performed a psychophysical experiment which attempted to shed light on the type of binaural neurons used by BiCI listeners to lateralize on the basis of ITD.

We measured the ability of four BiCI users to lateralize an interaural click as a function of long ITDs, many outside the range normally experienced by humans. The stimulus was a single biphasic pulse with durationof 54 μs/phase. The subject's task was to discriminate whether the click came from the left or the right. Percent correct lateralization was generally poor at ITDs of 250 μs or less; however it improved monotonically with ITD and each subject correctly lateralized at near 100 percent when the ITD reached 1000 μs. Beyond 1000 μs, lateralization remained at near 100%up to ITDs of from 2000 to 4000 μs. Above these ITDs, the percent correct for lateralization began to drop.

To investigate why lateralization performance degraded for ITDs above 2000 μs, theBiCI users were tested in a different task. While listening to the same stimuli used in the lateralization task, subjects reported whether they heard one single, fused sound or two distinct sounds, one in each ear. For ITDs below 1000 μs, BiCI subjects always reported a single, fused sound. For larger ITDs, the subjects began to report two distinct sounds. The ITD at which fusion began to break down varied across subjects, ranging from 2000 μs to above 10,000 μs. One interpretation of these results is that subjects were able to lateralize accurately until the ITD became so large that separate distinct sounds were heard in each ear, at which point lateralization became difficult.

The ability of BiCI subjects to lateralize at ITDs up to and beyond 2000 μs suggests that, if lateralization is based on interaural coincidence detector neurons as proposed in the Jeffress model, then the best ITDs of some of these coincidence detectors must be very large to encode these large ITDs. An alternative model for ITD sensitivity uses EI neurons that receive excitatory inputs from one ear and inhibitory inputs from the opposite ear. Because this model is able to encode very large ITDs, it may better account for our psychophysical results than models relying on coincidence detection.

Supported by NIH Grants R01 DC005775 and P30 DC005209

T47: PERCEPTUAL BENEFIT OF A LOW-FREQUENCY ACOUSTIC SIGNAL WITH ELECTRIC STIMULATION FOR CHILDREN AND ADULTS

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Although cochlear implants (CIs) have dramatically enhanced the communication abilities of patients with hearing loss, there is still room for improvement. One idea that has been considered as a way to enhance speech recognition abilities is to amplify any low-frequency residual hearing patients may have and present that in combination with the electric signal they get through CIs. Potential roles of that low-frequency signal are: (1) It may provide additional linguistic information that could be useful in recognition, such as prosody and cues to word segmentation; (2) It could serve as a facilitator in the perceptual organization of the electric signal obtained through the CI.

Nonetheless, questions remain regarding the idea of combining electric and (minimal) acoustic signals. The study reported here sought answers for those questions. This was done by creating 4-channel vocoded versions of sentence (HINT) and word (CID) materials (250 - 8000 Hz), and presenting them alone (VOC stimuli) and in combination with the acoustic signal, low-pass filtered below 250 Hz (VOC+low stimuli) to adults (20) and 7-year-old children (20) in diotic and dichotic modes. In the dichotic mode, either VOC signals were presented to one ear, or the VOC signal was presented to one ear and the low-pass signal to the other. Four hypotheses were generated. H1: Recognition should be better for the VOC+low than for the VOC stimuli because the low-pass signal provides useful information and/or facilitates perceptual organization. H2: Recognition should be better in the diotic than dichotic condition because spectral integration would be more easily accomplished. H3: The benefit achieved with the combined signal would be greater for sentences than for words either because the type of information gained by the low-frequency signal best facilitates recognition at that level or long stretches of signal best promote perceptual organization. H4: Children would benefit more than adults, especially if the benefit is due to longer signals providing information about prosody and word segmentation because children likely rely on this kind of information more than adults.

Results supported the first three hypotheses, but not the fourth. Conclusions were that listeners benefit from the addition of an acoustic signal to electric stimulation, even if that signal is extremely low frequency. The magnitude of this benefit is best realized in diotic presentation, but may not be fully realized by tasks requiring only the recognition of individual words. Adults and children demonstrate benefit similar in magnitude, suggesting that the mechanism of effect may be that the acoustic signal facilitates perceptual organization of the electric signal.

This work was supported by Grant R01 DC000633 from the National Institute on Deafness and Other Communication Disorders.

T48: BIMODAL COCHLEAR IMPLANTS: DO WE NEED TO BALANCE THE LEVEL OF THE ELECTRIC AND ACOUSTIC SIGNALS?

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Bimodal cochlear implant patients, i.e., patients who have an implant in one ear and low-frequency hearing in the contralateral ear, generally achieve higher scores on tests of speech understanding than patients who receive stimulation from a cochlear implant. At issue in this project is the level of the acoustic signal relative to the CI signal that will produce the greatest benefit to speech understanding. We presented the acoustic signal at levels ranging from just over detection threshold to above the level of the CI signal and asked how performance varies as a function of the difference in level between the CI signal and the acoustic signal.

Five bimodal CI users who normally used a Cochlear Corporation signal processor were tested. For these tests, CI signal processing was implemented on a Personal Digital Assistant (PDA) platform. The platform allowed the transmission of electric and acoustic stimulation synchronously. The platform also allowed us to individually shape the stimulation delivered to the ear with low-frequency hearing to match NAL-PR targets.

The stimulus material was drawn from the AzBio sentence corpus. The noise was multitalker babble.

All five subjects received benefit from adding the acoustic signal to the CI signal. We found that (i) acoustic signals perceived as significantly softer than a CI signal can contribute to speech understanding in the bimodal condition, (ii) acoustic signals that are slightly softer than, or balanced with, a CI signal provide the largest benefit to speech understanding and (iii) acoustic signals presented at MCL provide nearly as much benefit (within 4 percentage points) as signals that have been balanced with a CI signal. The latter outcome may simplify bimodal fitting.

Support provided by a subcontract to the first author (Michael F. Dorman) from NIH grant R01 DC010494 (Philip Loizou). Professor Loizou passed away on July 22th, 2012.

T49: BINAURAL INTERACTION IN UNILATERAL COCHLEAR IMPLANT USERS WITH BILATERAL LOW-FREQUENCY RESIDUAL HEARING.

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Introduction
Bimodal hearing refers to the condition in which electric hearing via a cochlear implant (CI) is combined with residual hearing in the non-implanted ear. There is electrophysiological evidence that binaural integration of electric and acoustic signals occurs in these patients [Battmer et al. unpublished data]. The picture becomes more complex in the case of unilateral electro-acoustic stimulation (EAS). Unilateral electric hearing is then often combined with bilateral low-frequency residual hearing. This raises the question whether these patients only benefit from the presence of bilateral low-frequency information or if also acoustic-electric binaural interaction occurs. In a normative study, we found that binaural interaction can be investigated using 500 Hz tone-burst auditory brainstem responses (500 Hz TB ABR). Using a combination of 500 Hz TB ABR and electrically evoked ABR, acoustic-electric binaural interaction in EAS users with bilateral low-frequency residual hearing was investigated.

Method
Three CI-users implanted with the Nucleus Hybrid L24 implant participated in the study. All had bilateral low-frequency residual hearing. ABRs were recorded in three stimulus conditions: left, right and binaural stimulation. The non-implanted ear was stimulated using 500 Hz TB, the implanted ear was stimulated using biphasic pulses. A binaural interaction component is derived from the ABR (ABR-BIC) by subtracting the binaural response from the sum of both monaural responses. In addition, performance on binaural tasks was assessed by means of speech in noise tests, sound localization tests and a questionnaire.

Results
Electric-acoustic ABR-BICs from three EAS users will be presented. These electrophysiological data will be linked with results on behavioral measures.

Conclusion
It is hypothesized that electric-acoustic ABR-BICs will be present in EAS users with good performance on binaural tasks.

T50: BINAURAL SUMMATION OF SPECTRALLY SIDE-SEPARATED SPEECH AND BINAURAL FUSION OF TEMPORALLY ALTERNATING SPEECH IN COCHLEAR IMPLANT RECIPIENTS WITH UNILATERAL SEVERE TO PROFOUND HEARING LOSS

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Introduction:
In cochlear implant (CI) recipients with acquired unilateral severe to profound hearing loss and normal hearing or moderate hearing loss on the contralateral ear, the central auditory system receives signals from the electrically stimulated CI ear as well as from the acoustically stimulated contralateral ear, i.e. signals of different auditory modalities. In the present study the auditive perception and processing in CI recipients with asymmetric hearing loss is investigated with regard to binaural summation of spectrally side-separated speech and binaural fusion of temporally alternating speech.

Methods:
To date four patients provided with Nucleus CI systems have been included into the study. Binaural summation of spectrally side-separated speech was analyzed for words of the dichotic Feldmann [Feldmann, 1965, Arch OhrenNasenKehlkopfheilkd., 184, 294-329] test using a modified version of the binaural summation test according to Matzker [Matzker, 1958, Stuttgart, Thieme-Verlag]. Two conditions of spectral separation were investigated. In one condition the better ear was acoustically stimulated with low-frequency and the electrically stimulated ear with high-frequency speech bands of different bandwith, and in the other condition vice versa.
In terms of binaural fusion of temporally alternating speech, investigation was performed by applying a modification of the binaural alternating speech test according to Hennebert [Hennebert, 1955, ActaOto-Rhino-Laryngol Belg., 9, 344], Bocca [Bocca, 1960, V. Congr. Int. Soc. Audiol., Bonn, 63] and Calearo [Calearo, 1960, V. Congr. Int. Soc. Audiol., Bonn, 1960, 67] using Feldmann words. Binaural fusion of temporally alternating speech has been tested for 12 temporal alternation rates between 2.6 and 32 Hz.

In both investigations the dichotic speech discrimination as well as the monotic speech discrimination with the CI ear and the contralateral ear is measured, respectively.

Results:
All recipients showed superior dichotic speech discrimination of spectrally side-separated speech as well as temporally alternating speech compared to monotic speech discrimination with the CI ear and the contralateral better ear only, respectively.

Conclusion:
Presented preliminary results indicate that CI recipients with acquired unilateral severe to profound hearing loss and normal hearing or moderate hearing loss in the contralateral ear can summate the information in spectrally side-separated speech and fuse the information in temporally alternating speech, respectively.

T51: THE VIRTUAL REALITY AS A TOOL FOR THE INVESTIGATION OF THE LOCALIZATION WITH SINGLE SIDED DEAF COCHLEA IMPLANT USERS.

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The localization of sound sources based on acoustical features is limited with monaural hearing, as in unilateral cochlear implant users or single sided deaf listeners, in comparison with normal binaural hearing. In the clinical setup the localization is investigated in a stationary test setup where the subject points out the direction of motionless sound sources. Since most of these investigations covered the localization in quiet only the outcome of the localization under more realistic conditions like reverberated acoustics is unknown. Under these realistic conditions the relative movements between the listener and the sound source might additionally influence the localization abilities of the listener.

To measure the localization abilities under potentially more realistic but yet controlled conditions an alternative method of measurement is introduced which uses the capabilities of the virtual reality (VR). In the VR localization setup the participants of the study show the direction of the sound source and the position of the invisible sound source in an acoustically and visually simulated room by moving themselves through the virtual room to the localized sound source. The required length of the path through the VR and the time needed until the sound source has been found will be discussed as measure for the localization abilities.

Rooms with different reverberation times were simulated in the VR to test how the localization of cochlear implant (CI) users is affected by reverberation. With the localization in these different listening environments it was tested if single sided deaf CI users benefit from the integration of the acoustical and electrical hearing in favorable conditions only or in more challenging listening conditions as well. First localization results of unilateral and bilateral cochlear implant users will be presented. The results of a stationary localization test setup are compared to the VR localization setup.

T52: FREQUENCY DEPENDENT LOUDNESS BALANCING TO OPTIMIZE HEARING AID GAINS IN BIMODAL CI USERS

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Despite the overall positive results of the combined use of a cochlear implant (CI) and a contralateral hearing aid (HA), large individual differences in bimodal benefit have been reported. A possible reason for poor results is a mismatch in loudness between ears across the dynamic range and/or across frequencies.

At present, bimodal fitting guidelines recommend adjusting overall HA gain to equalize loudness of soft and loud speech in both ears, with limited adjustments of the frequency response on the basis of clarity judgments [1]. In the present study, we adjusted HA gain for soft and loud input signals in three frequency bands (0 - 500, 500 - 1000 and >1000 Hz) to match loudness with the CI ("three-band" balancing). Before balancing, HA gain was reduced where the hearing loss exceeded 120 dB HL. This procedure was compared to a simple balancing procedure using the International Speech Test Signal (ISTS) at 65 dB SPL ("broadband" balancing) and the HA manufacturer's prescriptive rule.

Fifteen CI users, all wearing an Advanced Bionics Harmony processor, were fitted with one and the same HA (Phonak Naida S IX UP), eliminating variability due to devices. In a two-visit cross-over design with four weeks between sessions, bimodal benefit using "broadband" and "three-band" balancing was tested with subjective questionnaires and speech understanding in quiet and noise under several listening conditions.

Preliminary speech in noise results (between-subject) suggest a bimodal benefit ranging from 1 dB (stationary noise, S0N±90) to 5 dB (ISTS noise, S0N±90 and S0NCI). Three-band balanced subjects performed better than broadband balanced subjects in ISTS noise (1 to 3 dB), but worse in stationary noise, suggesting that three-band balancing resulted in increased energetic masking due to higher low-frequency amplification but an improved ability to separate voices. Subjective questionnaires suggest a slight preference for three-band balanced fittings with respect to sound quality and daily life speech understanding.

Data collection is underway to correlate bimodal performance with patient characteristics including hearing thresholds, the residual dynamic range, and cochlear tuning.

This study is financially supported by Advanced Bionics.

[1] Ching, T. Y., van Wanrooy, E., & Dillon, H. (2007). Binaural-bimodal fitting or bilateral implantation for managing severe to profound deafness: A review. Trends in Amplification, 11(3), 161-192

T53: HEAD MOVEMENT BENEFIT FOR SOUND LOCALIZATION IN SIMULATIONS OF COCHLEAR IMPLANT AND ELECTROACOUSTIC STIMULATION PAIRED WITH CONTRALATERAL HEARING AID, COCHLEAR IMPLANT, OR ELECTROACOUSTIC STIMULATION

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Cochlear implant (CI) users' limited ability to access or use acoustical cues for sound localization can result in left/right (L/R) and front/back (F/B) errors. Even limited head movement is highly beneficial in reducing these errors in acoustically hearing listeners. In this study, we compared normally hearing listeners' sound localization performance with various simulated CI device configurations in tasks allowing either no, limited, or free head movement. We also measured interaural time- and level-difference (ITD and ILD) discrimination thresholds (JNDs) in each configuration.

Localization targets were presented over headphones in virtual auditory space with real-time head tracking and using each subject's behind-the-ear head-related transfer functions (BTE HRTFs) to create simulations of hearing aid (HA) and CI stimulation in various combinations. The low-frequency acoustic or HA component of the stimulation was a 0.5-1-kHz band of noise filtered by the BTE HRTFs. The high-frequency CI component was a 1-7 kHz band of noise filtered by BTE HRTFs altered to mimic the processing of a 6-band noise vocoder. Uncorrelated noise samples were used for left- and right-ear CI stimulation to model bilateral CI users' typically poor ITD sensitivity. For electroacoustic (EAS) simulation, HA and CI components were summed in the same ear. We tested four simulated device configurations: bilateral CI; bimodal (CI paired with contralateral HA, CI+HA); bilateral EAS; and EAS paired with contralateral HA (EAS+HA). ITD and ILD JND measurements used similar stimuli without HRTF filtering.

L/R errors were frequent under head-static conditions in configurations with asymmetrical stimulation (CI+HA and EAS+HA), and neither limited nor free head movement eliminated those errors in those configurations. F/B errors were frequent in all device configurations under head-static conditions. Near-normal head movement benefit (elimination of F/B errors for both limited and free head movement) was obtained only for the bilateral EAS configuration, which provided both symmetrical stimulation and access to low-frequency ITD cues. In the bilateral CI configuration, which was symmetrical but provided no access to low-frequency ITD, only free head movement eliminated F/B errors. In the EAS+HA condition, which was asymmetrical but did provide ITD access, neither limited nor free head movement eliminated F/B errors.

Across device configurations, ITD JNDs were strongly correlated with front/back error rates in the limited head movement localization task, but ILD JNDs were not. Bilateral EAS provided the lowest ITD JNDs, followed by bilateral CI. ITD JNDs were similar and very high for both EAS+HA and CI+HA even though EAS+HA should provide access to low-frequency ITD. The results suggest that both symmetrical stimulation and access to low-frequency ITD are necessary for high sensitivity to dynamic sound localization cues.

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T54: IMPROVED AESTHETIC RESPONSES TO MUSIC IN BIMODAL AND BILATERAL COCHLEAR IMPLANT USERS

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While cochlear implant (CI) users generally perform well in speech understanding tasks in quiet environments, much improvement is needed for non-speech sounds such as music. Unsatisfactory music perception with cochlear implants is due to three main factors: poor pitch perception, poor timbre discrimination, and degradation of auditory streaming ability. Auditory streaming is the ability to segregate and combine different auditory information streams and is mainly based on perceptual differences between sources. Unfortunately, the CI degrades the acoustic cues that give rise to perceptual differences between sound sources, thus reducing the ability of the CI user to segregate different sound sources. This in turn reduces the ability to separately hear multiple lines of melody and different instruments, as well as different voices among many.

Our previous research suggests that musical streams are easier to segregate when the pitch distance between them is large (Marozeau et al 2010, PLoS ONE; 5:6). In this study we used short piano pieces known to evoke happy or sad emotional responses in normally-hearing (NH) listeners (Vieillard, Peretz et al 2008, Cognition & Emotion; 22:4). We increased the perceptual distance between streams by sending the two parts of the pieces to different ears in 11 NH adult participants, or different hearing devices for 11 bimodal listeners (CI with hearing aid in the contralateral ear) and 8 bilateral listeners (CI in both ears) CI users. We recorded three ratings on dichotomous scales after each short piece was played: a happy/sad emotional rating, a like/dislike aesthetic rating and a clear/unclear stream segregation rating. We hypothesised that the emotional, aesthetic and stream segregation ratings would improve when the two parts were sent to separate ears compared to when they were combined in a single ear.

We found that on average, hearing-impaired participants successfully rated the happy pieces as happy, and the sad pieces as sad. Thus, CI users were able to discern the emotional content of simple piano pieces without additional separation of the melody and harmony. However, separating the parts did significantly improve stream segregation ratings for both hearing-impaired groups. We also found that hearing-impaired listeners preferred listening to happy pieces compared to sad pieces, while NH listeners made equal aesthetic ratings for happy and sad pieces. Further acoustic analysis of the differences between happy and sad pieces may help to explain this unexpected effect.

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T55: DO COCHLEAR IMPLANT USERS RELY ON DIFFERENT ACOUSTIC CUES THAN NORMAL-HEARING ADULTS? A STUDY OF CUE WEIGHTING AND SALIENCE

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It is widely accepted that speech signals contain highly redundant acoustic information. Adult listeners retain an exceptional degree of speech intelligibility even under highly degraded acoustic conditions. This principle has allowed the successful use of cochlear implant (CI) devices, which typically deliver a sensitive representation of amplitude cues but degraded spectral cues. It is unclear, however, whether amplitude cues are as effective as spectral cues in speech perception by CI users, and whether perceptual weighting strategies are dependent on the salience of these cues.

To examine cue weighting in CI users, the /ba/-/wa/ stop-glide contrast was chosen. For this contrast, either an amplitude cue (amplitude rise time, ART) or a spectral cue (formant rise time, FRT) can cue the distinction between /ba/ and /wa/. It is known that adult, normal-hearing English speakers strongly weight FRT and largely ignore ART in the /ba/-/wa/ distinction, even though both cues are salient. Because CI users should have good access to amplitude information but likely have poor access to spectral information, questions arise as to their cue-weighting strategies. This study examined: (1) whether adults with CIs retain typical weighting strategies or adjust them based on salience of cues; and (2) how any adjustments to weighting strategies affect word recognition.

Twenty adult English-speaking CI users participated. Subjects underwent testing on three tasks: labeling decisions along /ba/-/wa/ continua, discrimination testing along ART and FRT continua, and word recognition. Results revealed wide variability in cue-weighting strategies among subjects. Spectral cue weighting could be explained by salience of the FRT cue; the same was not true for ART cue weighting and salience. Spectral cue weighting explained variance in word recognition. Age of onset of hearing loss explained spectral weighting but not unique variance in word recognition.

Conclusions were made that adult CI users show significant variability in cue-weighting strategies, and those subjects for whom spectral salience and spectral cue weighting were greatest showed the best word recognition. Although speech signals may be highly redundant, not all cues within the speech signal are equally effective. These findings have significant implications for CI design and fitting to improve spectral cue salience, as well as the potential role of training to develop the most effective perceptual strategies.

This work was supported by Grant R01 DC000633 from the NIDCD.

T56: A NEW CLINICAL SPEECH PERCEPTION TEST FOR SAUDI CHILDREN WITH COCHLEAR IMPLANTS

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Objective: The purpose of this study was to develop an Arabic Lexical Neighbourhood Test both as a clinical assessment tool for Saudi children and to investigate the lexical effects of word frequency and neighbourhood density on the identification of early-acquired words in a language whose structure is very different to that of English.

Lexical effects have been examined in children with normal hearing both in noise and with vocoded speech simulating a cochlear implant. In addition, a test-retest reliability study has been performed.

Method: Lexically controlled words were created from language samples collected from children between the ages of three and five. Four word lists were generated, two ‘easy’ lists and two ‘hard’ lists. Each list included 25 items. The definition for ‘easy’ words and ‘hard’ words was adopted from the English Lexical Neighbourhood Test developed by Kirk (1995).
In the study of lexical effects, one group of 24 five year old normal hearing children were presented with the spoken words in a background of speech spectrum shaped noise at an SNR set for each child. A second group of 24 normal hearing children, also aged five, heard speech in the quiet processed through a noise vocoder in conditions with four and eight channels.
For the test-retest reliability study a further 45 children were tested twice, two to four weeks apart, with speech in noise at an individually set SNR.

Results: In the lexical effects study, children scored significantly higher on lexically ‘easy’ words than on lexically’ hard’ words. Performance was significantly correlated with neighbourhood density but not with word frequency. An increase of vocoder channels from four to eight led to a significant increase in performance for both ‘easy’ and ‘hard’ lists.

In the test-retest reliability study, the correlations for the ‘easy’ words and the ‘hard’ words were calculated separately. The ‘easy’ word scores in the first and second session were significantly and positively correlated ($r=.721$, $p<0.01$) as were the ‘hard’ word scores ($r=.590$ and $p<0.01$). The mean absolute difference in children’s scores from the first session to the second session was 5.8% for the ‘easy’ words and 8.8% for the ‘hard’ words and this is within the binomially estimated random variability.

Conclusion: The Arabic Lexical Neighbourhood Test is a reliable measure for speech perception in Saudi Children. Despite Arabic phonology being substantially different to that of English, the findings support the view that Arabic words are organized in the lexicon on the basis of neighbourhood density and word frequency as proposed by the Neighbourhood Activation model.

T57: IDENTIFICATION OF VOWELS IN /DVD/ SYLLABLES BY COCHLEAR IMPLANT USERS: RELIANCE ON VOWEL DURATION CUES

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Donaldson et al. (CIAP 2011) examined vowel identification performance in cochlear implant (CI) users for full /dVd/ syllables and for partial /dVd/ syllables that retained only the vowel edges (dynamic spectral cues) or only the vowel centers (quasi-static spectral cues) with duration cues removed. Performance was substantially poorer for the partial syllables than for the full syllables. An analysis of vowel errors suggested that the poorer performance for partial syllables could stem from a strong reliance on vowel duration cues in the full-syllable condition. The present study examined the possibility that CI users rely more strongly on vowel duration cues than young, normal-hearing (YNH) listeners by testing vowel identification for partial /dVd/ syllables in which vowel duration cues were either removed or preserved.

Subjects were 10 YNH listeners and 11 adult CI users (14 ears). Stimuli were naturally-spoken /dVd/ syllables recorded from three female talkers, and partial syllables derived from the same stimuli to create 6 additional stimulus conditions: three syllable types (center-only; edges-only with 40-ms edges; and edges-only with 20-ms edges) x two duration conditions (preserved; removed). For the center-only syllables, duration cues were preserved by maintaining vowel durations that were proportional to the durations present in the original syllables, or were removed by excising a fixed duration (80 ms) of the vowel center. For the edges-only syllables, duration cues were preserved by maintaining the original vowel duration, or were removed by adjusting the duration of the silent gap to be equal to the average vowel duration of all syllables. For the center-only condition, duration-preserved stimuli were associated with significantly higher vowel-identification scores than duration-removed stimuli in both groups of subjects. However, the average effect of duration cues was larger for the CI group than for the YNH group, primarily because a few of the CI users exhibited very strong effects of duration. The presence or absence of duration cues had no significant effect on vowel-identification scores for the edges-only conditions in either group of subjects. However, some individual CI users exhibited a clear improvement in scores for the duration-preserved stimuli as compared to the duration-removed stimuli.

Overall, findings suggest that vowel duration cues have a relatively small influence on vowel identification performance in YNH listeners and in some CI users. However, a subset of CI users appear to rely strongly on vowel duration cues. A strong reliance on vowel duration cues could be a natural consequence of poor spectral acuity; it could also reflect a perceptual weighting scheme that was established prior to implantation when the listener had limited access to spectral cues.

Donaldson, G.S., Rogers, C.L., Russell, B.A. and Cardenas, E.S. (2011) Contributions of static and dynamic cues to vowel identification by cochlear implant users.Conference on Implantable Cochlear Prostheses, Asilomar, CA.

T58: THE EFFECT OF MICROPHONE LOCATION ON SPEECH INTELLIGIBILITY AND LOCALIZATION IN UNILATERAL CI USERS

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Advanced Bionics recently introduced the TComm, an accessory for the Neptune and PSP sound processors. TComm allows access to the T-Mic, a microphone placed at the opening of the ear canal. The traditional microphones of the Neptune and PSP sound processors are integrated into the headpiece, which is positioned above and slightly behind the pinna. Thus, using the TComm together with the T-Mic is intended to optimize acoustic performance, due to the more natural microphone placement for Neptune and PSP speech processors. Previous studies have shown an advantage for Auria and Harmony users with the T-Mic, in terms of better speech understanding in noise, improved localization and better subjective listening.

The effect of microphone location was evaluated for the TComm together with the T-Mic, as well as for the headpiece microphone. Physical measurements were made with the Neptune processor, which is also being used in an ongoing study involving unilateral CI users.

Polar patterns for both microphones were measured with a KEMAR, showing different behaviors for direction and frequency. The polar pattern of the T-Mic showed a more directional response than the headpiece microphone. The directivity indices measured in an anechoic chamber differed by approximately 2.5 dB between microphones: indicating a directivity advantage for the T-Mic relative to the headpiece microphone. However, the improvement in directivity index for the T-Mic was smaller in the (low-reverberant, sound-damped) room in which the measurements with CI users were performed. Nevertheless, there was still an advantage of approximately 1 dB, which we expected to observe in the study with CI users.

Performance was measured with unilateral CI users, for both microphones relative to the Harmony with the T-Mic configuration. The CI users were fitted with the Neptune processor together with either, the TComm and T-Mic, or the headpiece microphone in a cross-over study design. Each microphone was used during a take-home phase of 4 weeks and afterwards, localization and speech intelligibility in noise were measured in a setup with eight and four loudspeakers respectively.

In total 10 CI users will be enrolled in the study, currently 4 were recruited. Preliminary speech intelligibility results showed a benefit of approximately 3 dB for the T-Mic compared to the headpiece microphone, which is in line with what was predicted from the physical measurements.

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WEDNESDAY POSTER ABSTRACTS

W1: EFFECT OF MASKER LEVEL, RATE, AND SPATIAL SEPARATION ON PSYCHOPHYSICAL THRESHOLDS FOR INTERLEAVED PULSE TRAINS

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Speech-processing strategies that utilize relatively high-rate stimulation can produce temporal and spatial interactions. Temporal interactions may produce either forward-masking or facilitative effects, depending on the timing and relative level of the sequential pulses. Spatial interactions are expected to decrease with increased distance between electrodes that deliver sequentially timed stimuli. It is not clear how these temporal and spatial interactions vary as a function of stimulation rate. The purpose of this study was to assess the effects of masker level, stimulation rate, and spatial separation on perceptual detection of interleaved pulse trains.

Masker and probe stimuli each consisted of 300-ms pulse trains presented to adjacent electrodes in the middle of the array. Stimuli were interleaved using an 8-usec interval between the offset of each masker pulse and onset of each probe pulse. This paradigm was chosen over a traditional forward-masking paradigm because it better represents sequential stimulation typically used in CIs. For the level and rate portions of the experiment, two rates were tested (1700 pps and 3400 pps) and four masker levels ranging from sub-threshold to “loud but comfortable” (rating of “7” on a 10-point scale) were used. For the spatial portion of the experiment, the masker was fixed on the most apical electrode and the probe was incrementally positioned along the array. An adaptive, 3IFC task was used to determine probe threshold for each masked condition.

Preliminary data from 7 Nucleus recipients show no significant differences between probe thresholds obtained with no masker (probe quiet condition), a sub-threshold masker, or a masker at threshold. This suggests that maskers with a short temporal offset presented at levels below or near threshold do not produce a facilitative effect, which was not consistent with our hypothesis. The amount of masking increased as masker levels increased from loudness ratings of “5” to “7”, as expected. Based on our earlier ECAP data, we expected smaller masking effects for the 3400-pps train because the faster rate would presumably yield a more stochastic state than the 1700-pps train, and would therefore produce less masking. Results from this preliminary data set suggest that masking trends were the same for both stimulation rates. Finally, significant masking effects remained at the widest masker-probe separation (masker: E22, probe E2) for interleaved stimuli.

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W2: ADAPTIVE ELECTRODE SELECTION USING PSYCHOPHYSICAL MEASURES

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Previous research suggests that electrode selection based on psychophysical measures can result in improved speech recognition [Zwolan, Collins, and Wakefield, 1997, J. Acoust.Soc. Am., 102, 3673-3685; Boéx, Kós, and Pelizzone, 2003, J. Acoust.Soc. Am.,114, 2058-2063; Garadat, Zwolan and Pfingst, 2012, J. Acoust.Soc. Am., 127, 977-987]. These studies excluded the electrodes with the highest potential to negatively impact speech recognition, and this resulted in a single reduced electrode set. However, they did not exhaustively investigate the effect that varying the number of excluded electrodes could have on speech recognition. Such a process would involve testing speech recognition for each distinct set of electrodes, which would require many testing hours. Here, a more time efficient method for psychophysics-based electrode selection is proposed that adaptively varies the electrodes included in a listener's map according to the responses to a speech recognition task.

In this study, electrode discrimination, forward masking and modulation detection measures were collected for all active electrodes in each subject's map. For each metric, the data was used to rank electrodes according to their potential to negatively impact speech recognition. Distinct perceptually-defined electrode sets were determined by excluding an increasing number of electrodes with the highest potential to confound. Each of these sets was assigned a prior distribution for the probability of discrimination in a closed-set speech recognition task. After the presentation of a single speech token using a given electrode set, the distribution corresponding to that set was updated using a Bayesian framework. In order to determine the electrode set to test next, the expected information gain [Kastella, 1997, IEEE Trans. Syst., Man, Cybern. A, Syst. ,Humans, 27, 112-116] was calculated for each of the electrode sets. The set with the highest expected information gain was then used to present the next speech token. This process was repeated until it converged on a final electrode set. The performance of the final electrode set was validated on a speech recognition task. Electrode selection using this adaptive method could significantly reduce the time needed to determine the set of electrodes best suited for a particular listener. Further, this approach is not restricted to electrode selection optimization and may provide a time-efficient method of optimizing any of the cochlear implant speech processing parameters.

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W3: THE ROLE OF AUDITORY FEEDBACK ON VOCAL PITCH PRODUCTION IN COCHLEAR IMPLANT USERS

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Cochlear implants (CI) have been highly successful in improving speech perception and production in people with profound hearing loss. However, poor perception of pitch and distorted auditory feedback through CI devices may negatively impact the accuracy of vocal pitch production in CI users. In this study, we investigated two questions. First, how does pitch perception relate to vocal pitch production in CI users? Second, how does real-time CI-mediated auditory feedback influence vocal pitch production in CI users?

We recorded the vocal production in postlingually-deafened CI users of 1) vocal pitch matching for pairs of synthesized syllable /ma/ in the pitches of seven musical notes; 2) spontaneous singing of familiar melodies; 3) reading of consonants (in the form of /aCa/ syllables) and vowels (in the form of /hVd/ and /sVt/ syllables). The fundamental frequency (F0) of vocal production and pitch contour in CI users was evaluated and compared to results of vocal pitch production in NH controls. To assess the impact of CI-mediated auditory feedback on vocal pitch production, we compared singing and production of consonants and vowels in two conditions (CI-On vs. CI-Off).

Our results demonstrate that pitch production of CI users is significantly less accurate than that of NH controls. CI users typically produced lower vocal pitch with greater variation in comparison to NH controls. However, the performance of pitch contour produced by CI users as judged by NH listeners was relatively accurate and was significantly correlated to their pitch contour perception. Vocal pitch production during the CI-Off condition was generally less stable than during the CI-On condition. In particular, the fundamental frequency of CI user's vocal production was typically shifted towards higher frequencies during CI-Off in comparison to CI-On.

These findings show that vocal production of CI users captures the overall contour but not the absolute pitch of the target sounds, indicating that vocal production of CI users is affected by the impoverished auditory feedback received and the poor pitch perception observed in most CI users. Furthermore, these findings suggest that optimizing auditory feedback in CI users should improve both vocal pitch production and speech intelligibility.

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W4: TEMPORAL PATTERN PERCEPTION USING MONOPOLAR AND TRIPOLAR STIMULATION

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The question of whether cochlear implant users can perceive temporal structure is fundamental in the design and implementation of novel speech coding strategies. F0 information is represented by modulations at the output of the speech processor. Cochlear implant listeners have difficulty in hearing the pitch of the F0 modulations due to the modulations on different electrodes being out-of-phase, and also the overlap of excited neurons due to current spread within the cochlea. If greater independence were possible between electrodes, this may result in improved pitch perception.

Research has suggested that tripolar (TP) stimulation may lead to an increase in electrode independence than presently occurs using monopolar (MP) mode. The experiments described here investigated the amount of electrode independence by examining the effect of mode (MP or TP) on temporal pattern perception using single- and dual-electrode stimuli. Single-electrode testing examined listeners' ability to discriminate between different interpulse intervals, and dual-electrode testing examined the ability to discriminate between pulse trains with different inter-electrode timing. The hypotheses were a) that a narrower mode of stimulation would result in poorer transmission of temporal patterns than a wider mode on a single electrode, and b) that a narrower mode would be more advantageous in transmitting independent temporal information to different electrode positions.

Nine Advanced Bionics cochlear implant users were recruited. A partial tripolar configuration of 0.8 was used in all participants. Results showed no effect of mode on temporal perception at either a single electrode place or across electrodes. This suggests that either the width of neural activation may not be very different in TP and MP modes at equal loudness levels, or the width of neural activation is not important for discriminating interpulse intervals. Inter-subject variability was noted, and this may suggest that narrow modes are not helpful for every implant listener. It may be that targeted use of narrow modes will be useful in listeners whose electrode-neuron interface is shown to be good.

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W5: LINKING TEMPORAL MASKING AND SPEECH UNDERSTANDING IN HEARING WITH COCHLEAR IMPLANTS

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Normal hearing (NH) listeners are very good at detecting a signal in a brief gap between two maskers, an ability that is paramount for understanding speech in a modulated background such as an interfering speaker. In NH, masking from the preceding (forward, FWD) masker lasts longer than masking from the succeeding (backward, BWD) masker. However, cochlear implant (CI) users struggle to detect signals in masker gaps, even though FWD masking behaves similar to NH (Shannon, 1990). We tested temporal masking from FWD, BWD and the combination of FWD and BWD maskers with CI users to investigate its impact on the ability to detect signals in the masker gaps and to relate it to speech understanding.

Detection thresholds of a 10 ms-long probe presented before, after, or in-between 300 ms-long constant-amplitude, constant-rate electric pulse trains were measured at the mid-array electrode of the CI with seven adult CI users. Generally, probe thresholds decreased with increasing delay or gap duration. However, the recovery from masking varied across participants from very little to full recovery at 100 ms delay. The recovery from BWD masking was slower than that observed in NH for all but one CI user, and was as slow as the recovery from FWD masking for several CI users. Recovery from masking from a combined FWD and BWD masker required gap durations longer than the delays needed for recovery from FWD masking.

We also measured speech reception thresholds (SRTs) in modulated noise of varying gap duration. In general, participants who showed a faster recovery from backward masking also showed lower SRTs, in agreement with the hypothesis that they were able to better listen into masker gaps. The results suggest that the difficulties CI users experience with understanding speech in modulated noise are related to the prolonged recovery from BWD masking.

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W6: EQUIVALENT MASKING PATTERNS OF BIPOLAR PROBES FORWARD-MASKED BY MONOPOLAR OR BIPOLAR MASKERS

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This research employed a forward-masking paradigm to estimate the current spread of monopolar (MP) and bipolar (BP) maskers, with current amplitudes adjusted to elicit the same loudness. Since the spatial separation between active and return electrodes is smaller in BP than in MP configurations, the BP current spread is more localized and presumably superior in terms of speech intelligibility. Because matching the loudness requires higher current in BP than in MP stimulation, previous forward-masking studies show that BP current spread is not consistently narrower across subjects or electrodes within a subject.

The present forward-masking measures of current spread differ from those of previous studies by using the same BP probe electrode configuration for both MP and BP masker configurations, and adjusting the current levels of the MP and BP maskers so as to match them in loudness. With this method, the estimate of masker current spread would not be contaminated by differences in probe current spread. Forward masking was studied in four cochlear implant patients, two females and two males, with speech recognition scores higher than 50%; that is, their auditory-nerve survival status was more than adequate to carry out the experiments.

The data showed that MP and BP masker configurations produce equivalent masking patterns (and current spreads) in three participants. A fourth participant displayed asymmetrical patterns with enhancement rather than masking in some cases, especially when the probe and masker were at the same location.

W7: IMPROVING PLACE-PITCH IN THE APEX VIA MANIPULATIONS IN PHASE DURATION

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Many cochlear implant users have difficulty in ordering electrodes according to their place-pitch, and confusions or reversals tend to be disproportionately prevalent in the apical region of the cochlea for deeply inserted electrode arrays (e.g., Boyd, 2011). Ambiguities in pitch associated with apical electrodes may arise at least in part due to undifferentiated stimulation of spiral ganglion neurons (SGNs) in the terminal bulb, in contrast to stimulation of the peripheral processes that project from the SGNs to the apical part of the Organ of Corti. Given the finding that low-amplitude, long-duration pulses can recruit smaller diameter fibers compared to relatively high-amplitude, short-duration pulses (Ranck, 1975), it may be possible to recruit surviving peripheral processes in the apex using alternative waveforms such as the low-amplitude, long-duration pulses.

In this study, six subjects were asked to pitch rank loudness-balanced, single-electrode stimuli that varied systematically in phase duration (PD) between subjects' clinical PDs (15-50 μs/phase) and 400 μs/phase. PD was manipulated on at least one apical, one middle, and one basal electrode for each subject. A significant decrease in mean rank (pitch) was observed with an increase in PD for all six subjects tested for apical electrodes ($p<0.05$). Changes in PD also produced significant effects on mean ranks in other regions of the cochlea; however, the directions of the changes were not consistent.

Elimination or amelioration of pitch reversals and confusions with increased PDs may improve speech or music reception with multichannel cochlear implants. We plan to investigate this possibility in future work, and the results will be reported at the conference.

W8: UNDERSTANDING THE EFFECTS OF CENTRAL AND PERIPHERAL MASKING IN COCHLEAR IMPLANT USERS

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Whereas cochlear implant (CI) listeners are able to comprehend speech in quiet environments, they are unable to parse relevant speech from noisy, acoustic scenes. Evidence from normal-hearing (NH) listeners suggests that both peripheral, cochlear encoding and central, brain processing are critical for speech-in-noise listening. It is well-established that wide peripheral filters contribute to poor segregation, however, the contribution of central processing in the electrically-stimulated brain is less well-explored. Using non-speech stimuli with and without spectral signal-masker overlap will allow us to determine peripheral vs. central contributions to parsing a masked signal.

Peripheral filter bandwidth was measured in 5 postlingually, deafened implantees with Nucleus devices using electrically evoked compound action potential forward masking techniques (Custom Sound software). All stimuli were presented through a research interface (HEINRI) at 300 pulses per second. The signal was fixed on electrode 11 while the masker was presented on four other electrodes placed either inside (energetic masking) or outside (informational masking) of the peripheral filter. Both signal and masker were 40 ms bursts, pulsed 4 times at 5 Hz. Masker electrode positions varied or were constant among bursts, and masker pulses were synchronous or asynchronous re signal pulses. Masker level was roved by ±3 clinical units to minimize loudness cues. A two-interval, alternative forced choice paradigm was used to measure signal detection thresholds in the presence of these maskers. We evaluated differences in electric vs. acoustic hearing by comparing thresholds measured from CI, NH, and CI simulated listening.

The results indicated: (i) central masking was present; (ii) on a population level, CI users did not show a significant threshold difference between informational maskers with varying or constant electrode positions, although, individual subjects could show small improvements in thresholds provided by maskers with a spectral cue; (iii) energetic maskers yielded lower detection thresholds than did informational maskers (most likely due to loudness cues and differences attributed to electric stimulation); and (iv) thresholds for both CI and NH subjects were lower in the asynchronous than in the synchronous timing condition. To examine a potential effect of bandwidth, we performed a CI acoustic simulation using 22 noise bands of varying bandwidth logarithmically-spaced between 0.2-14 kHz and measured detection thresholds in NH subjects. NH and CI simulation data indicated that restricting frequency resolution decreased the benefit of signal detection in a masker background containing a spectral cue.

We conclude that limited peripheral encoding in CI listening greatly reduces the benefit from central processes and effectiveness of cues for segregating a signal from masking. Future auditory prostheses or stimulation strategies that restore higher, peripheral frequency resolution may improve implant speech perception in noise.

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W9: ACROSS-ELECTRODE PROCESSING IN COCHLEAR IMPLANT USERS

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Cochlear Implant (CI) users usually perform well in quiet listening situations. In fluctuating interference however, performance is strongly reduced. The normal-hearing auditory system has elaborated strategies to deal with mixtures of fluctuating sounds. One such mechanism is based on correlated patterns of intensity changes across the spectrum. This common feature of natural sounds introduces binding cues for auditory stream formation. Comodulation Masking Release illustrates how such coherent fluctuations can improve signal detection. Our prior experiments with controlled direct stimulation in CI users showed significant CMR-magnitudes in a subgroup of CI users.

Therefore, we applied a flanking band type of experiment processed by a modified ACE-strategy (Zirn et al., 2013). The underlying CMR definition applied was the uncorrelated-comodulated difference [CMR(UM-CM)]. In our current work we are approaching the reference-comodulated CMR (RF-CM) definition. Due to the limited dynamic range involved in electric stimulation, amplitude mapping is a critical step using this CMR-definition: to deal with loudness summation due to additional flanking bands added to an On-Signal Masker, we set the RMS level in the CM condition to 70 % of the individual dynamic range of the listener. Starting from here, we used a 3-AFC paradigm to determine equal loudness of the stimuli: (i) On-Frequency Masker alone (reference condition, electrode 14) and (ii) On-Frequency Masker plus four flanking bands addressed to adjacent (18,16,12 and 10) or remote (22,20,8 and 6) CI electrode contacts. All experiments were conducted with the Nucleus Implant Communicator, which provides precise control of the stimuli. First results align with our prior CMR(UM-CM)] outcomes: the addition of comodulated flanking bands to an on-frequency masker lead to a release of masking in a subgroup (~30%) of CI users. In this subgroup, we measured CMR (RF-CM) magnitudes of 7 dB±4 with adjacent electrodes addressed and 4 dB±3 with remote electrodes addressed. This finding corroborates our notion that across-electrode processing enhances perceptual formation of auditory objects in electrically induced hearing.

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Zirn, S., Hempel, J. M., Schuster, M., Hemmert, W. (2013).Comodulation Masking Release induced by controlled electrical stimulation of auditory nerve fibers. Hear Res, 296, 60-66.

W10: SINGLE- AND MULTI-CHANNEL MODULATION DETECTION BY COCHLEAR IMPLANT USERS

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Modulation detection has been used to characterize cochlear implant (CI) users' temporal processing. While detection of modulation presented on a single channel has been extensively studied, little is known about multi-channel temporal processing. How does modulation sensitivity on single channels contribute to the multi-channel percept? Do CI users integrate stimulation on all channels or attend to only the best (or worst) channel? How does the loudness summation associated with multi-channel stimulation affect modulation detection?

In this study, detection of modulation presented to a single channel or to multiple channels was measured in 9 CI users. Experimental electrodes were evenly distributed from the base to apex. The stimulation rate for each electrode was 500 pulses per second. Modulation detection thresholds (MDTs) were measured at relatively low and high presentation levels and at relatively low (10 Hz) and high (100 Hz) modulation frequencies. Single- and multi-channel channel MDTs were measured at equally loud presentation levels multi-channel MDTs were also measured without compensation for loudness summation. Multi-channel MDTs were also measured with coherent modulation applied to 1 out of 4 channels or to all 4 channels

MDTs were generally better at the higher presentation level and at the lower modulation frequency. Differences in single-channel MDTs across stimulation sites ranged from 1.78 to 15.56 dB. At a fixed loudness level, multi-channel MDTs were generally poorer than single-channel MDTs, most likely due to the lower stimulation levels needed to compensate for loudness summation. Without the loudness compensation, multi-channel MDTs were most comparable to the best single-channel MDT.MDTs were similar when modulation was applied to a single channel or to 1 of 4 channels (the same channel in both conditions), suggesting that there was little interference among the 4 channels. MDTs were better when modulation was applied to all 4 channels, rather than only 1 of 4 channels, suggesting that subjects benefitted from coding modulation with a greater number of channels.

The present data suggest that for some CI users, multi-channel modulation sensitivity may be largely driven by the best component channel. For CI users with less across-site variability in MDTs, multi-channel stimulation may enhance modulation sensitivity. Because component channel stimulation levels must be reduced to compensate for multi-channel loudness summation (as in clinical fitting), modulation sensitivity is reduced for individual channels. As such, selecting only channels with good modulation sensitivity may reduce summation and allow forhigher stimulation levels on component channels, which may improve overall modulation sensitivity.

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W11: EXTRACTING INTERAURAL TEMPORAL INFORMATION FROM THE ENVELOPES OF LOW- AND HIGH-FREQUENCY MODULATED SOUNDS

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Normal-hearing listeners show a remarkable ability to hear out individual talkers against a background of competing voices, or in large rooms or halls where walls and other hard surfaces generate reflections of greater intensity than the original source. One factor that appears critical to such 'cocktail-party' listening is the ability to extract information from the temporal fine-structure (TFS) of sounds, particularly for frequencies below 1500 Hz. The vast majority of real-world sounds are strongly modulated in amplitude - human speech contains amplitude modulation (AM) rates ranging from those of syllables and phonemes to those responsible for conveying information about voice pitch (i.e. from 2 Hz up to about 300 Hz). Such modulations act as potent grouping cues, enabling individual talkers to be heard against interfering voices.

Here, using amplitude-modulated binaural beats, we investigate the interaction of TFS and temporal envelope at the perceptual and neural levels. Binaural beats, generated by low-frequency tones of slightly different frequency presented to the left and right ear, lead to an intracranial percept that oscillates along the interaural axis. Consistent with the notion of binaural sluggishness, however, a noticeable motion percept is apparent only for tonal binaural beats with rates below about 2 Hz; at higher rates the motion percept is lost, and the intracranial image is rendered static and diffuse with a characteristic roughness but no distinct lateralization. Through a combination of human psychophysics and brain imaging, however, we demonstrate that binaural sluggishness is overcome by a form of acoustic 'glimpsing' triggered by short rising segments in the amplitude of the sound, whereby ITD information during the remainder of the cycle is suppressed. Under reverberant conditions, where late-arriving spatial information may be spurious as to the location of the source, weighting binaural information more heavily towards the minima of the amplitude envelope favors accurate localization. Since most cochlear-implant (CI) processing strategies discard TFS information completely (and it remains uncertain the degree to which low-frequency binaural pathways are effectively stimulated in CI patients), we also examined the situation where the binaural beat is conveyed only in the modulated envelope of transposed tones with high-frequency (4 kHz) carriers, with second-order envelope modulated imposed diotically, mirroring the low-frequency case. Whereas ITDs conveyed in the TFS of pure tones centred at 250 Hz were weighted half-way up the slope of the modulation envelope (90°), second-order AM binaural beats centered at 250-Hz modulation rate were weighted at the energy peak (170° in the modulation cycle). Since, under noisy listening conditions, interaural information at the energy peaks is usually less reliable, this potentially helps explain the poorer performance of CI users in listening tasks that rely on them extracting binaural temporal cues in cocktail-party listening.

W12: PLACE AND RATE DEPENDENCY OF CHANNEL INTERACTION WITH SIMULTANEOUS STIMULATION

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Simultaneous monopolar electrode stimulation results in adirect summation of electrical fields and produces spectral smearing and loudness summation. Apreviously presented Channel Interaction Compensation (CIC) algorithm has been shown to effectively eliminate at least effects of loudness summation. CIC is based on an exponential model of spatial potential distribution within the scala tympani. Two constants, α and β, model the field decay towards the apex (α^n) and base (β^n) away from each electrode. A third constant, γ, models the decay of place-dependent field potentials from apex to base. CIC computes reduced pulse amplitudes such that aftersuperpositionthe field potentials at each electrode correspond to the sequential non-overlapping case.

In an ongoing study, CIC model constants are determined such thatfor unmodulated pulse trains, compensated simultaneous stimulation produces the same loudness assequential stimulation.Parametric variations include place and number of adjacent electrodes being stimulated, in addition to pulse rate. In three CI users so far, bursts of simultaneous and sequential stimuli were presentedalternately. CI listeners adjusted the loudness of the simultaneous stimulusto that of the sequential stimulus by clicking on screen buttonsforincreasing or decreasing CIC parameters (with $\alpha = \beta < 1$, hence $\gamma = 1$), making the simultaneous stimulus softer or louder, respectively.

Preliminary results show a highly significant effect of both place of stimulation and pulse rate on CIC model parameters for matched loudness (two-way RM ANOVA, $p < 0.001$). Mean values for α and β ranged from 0.57 for basal electrodes at 1500 pps to the maximum value of 0.88 for apical electrodes at 100 pps.

Results indicate a strong dependency of the effects ofcurrent spread on both place and rate of stimulation. Lower simultaneous pulse rates result in larger loudness summation than higher pulse rates. Spread of excitation is broader for apical than for basal electrodes, as reflected in larger interaction parameters α and β for apical electrodes. On the other hand, pitch ranking experiments have shown that in contrast to simultaneous stimulation, temporal channel interaction for low-rate sequentialapical stimulation is much lower than anticipated. Temporal codes can still be transmitted largely independently on adjacent apical electrodes.

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W13: COMPARISON OF TEMPORAL INTEGRATION AND MULTIPULSE INTEGRATION FUNCTIONS IN HUMANS WITH COCHLEAR IMPLANTS

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Previous work in guinea pigs with cochlear implants found that psychophysical temporal integration (detection thresholds as a function of stimulus duration for fixed-rate pulse trains) and multipulse integration (detection thresholds as a function of pulse rate for fixed-duration pulse trains) were predictive of the health of the implanted cochlea (Pfingst et al., JASA, 2011, 130:3954-3968), suggesting some commonality in the mechanisms underlying these two functions. In the current study, the relationship between these two functions was further compared in humans with cochlear implants.

The two functions were compared in 12 subjects at two stimulation sites. Psychophysical detection thresholds were measured using a four-alternative forced-choice (4AFC) adaptive procedure. Temporal integration was assessed using a pulse rate of 640 pps and pulse trains of 31.25, 62.5, 125, and 250 ms in duration. Multipulse integration was assessed using a pulse train duration of 250 ms and pulse rates of 80, 160, 320, to 640 pps. These four conditions for the two functions resulted in delivery of 20, 40, 80, or 160 pulses per stimulus.

Thresholds of the temporal integration functions were lower than those of the multipulse integration functions below 640 pps. However, temporal integration tested subsequently at lower rates (i.e., 320 and160 pps) resulted in a shift of the functions to higher thresholds. The slopes of multipulse integration were comparable to those of temporal integration for integrating the same number of pulses up to 320 pps, but the multipulse integration slopes became steeper between 320 and 640 pps.

Results were consistent with the “multiple looks” model in which short-term integration of auditory stimuli occurs in a time window of ~3 ms and the integrated information is then combined within a long-term window of ~300 ms (Viemeister and Wakefield, JASA, 1991, 90:858-865). Lower thresholds can result from increasing the number of pulses in a fixed time window (multipulse integration) or increasing stimulus duration at a fixed pulse rate (temporal integration). Applying to both functions, thresholds measured with pulse rates below ~330 pps involves long term integration that is less efficient than short-term integration whereas detection at pulse rates above ~330 pps involves both long- and short-term integration due to more than one pulse appearing in the short term (3 ms) integration window. Results of the study indicate that the multipulse integration and temporal integration functions are two similar processes that could both reflect the short- and long-term integration of auditory stimuli. This work, coupled with the work in guinea pigs, establishes a basis for using temporal integration or multipulse integration data for assessing the functionality of individual stimulation sites and identifying sites for selection or rehabilitation when fitting speech processors.

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W14: ASSESSING THE HEALTH OF THE IMPLANTED COCHLEA WITH NONINVASIVE MEASURES

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Biological conditions in the implanted cochlea are hypothesized to affect cochlear implant function. This has motivated research and clinical efforts aimed at preservation and restoration of normal cochlear biology. However, it is often difficult to determine the conditions of the cochlea at the time when cochlear implant function is assessed. Noninvasive measures for determining cochlear conditions are needed in order to assess the health of the cochlea throughout implant use.

We evaluated the relative abilities of four noninvasive input-output measures to predict levels and features of the health of the implanted cochlea. We measured slopes for two types of electrophysiological amplitude-growth-functions, electrically-evoked compound action potentials (ECAPs) and electrically-evoked auditory brainstem responses (EABRs), and two types of psychophysical functions, multipulse integration (threshold vs. pulse rate) and temporal integration (threshold vs. pulse-train duration).

To create a large range of conditions in the implanted cochleae of adult male guinea pigs we used various treatment procedures: 1. Implantation in a normal hearing ear, usually preserving some acoustic hearing; 2. Deafening with neomycin followed by inoculation with an adeno-associated viral vector (AAV) containing an active neurotrophin gene (NTF-3 or BDNF) to promote SGN survival and neurite regeneration, resulting in loss of all or most hair cells, and SGN densities from low to moderate; and 3. Deafening with neomycin, and in some cases inoculating with an empty AAV, resulting in loss of all hair cells and degeneration of most SGNs. Analysis approximately one year following these treatments revealed large differences across animals in the amount of surviving inner hair cells (IHCs), the density of spiral ganglion neurons (SGNs) in Rosenthal's canal, the density of peripheral processes (PPs), the level of ensemble spontaneous neural activity (ESA), and the amount of fibrous tissue and new bone surrounding the implants.

Good multipulse integration and temporal integration were found only in cases with the best cochlear health. I.e., the slopes of these functions were steep only in animals that had surviving IHCs, >50% SGN density, and moderate to good ESA levels. The slopes of these functions were not different for cases with poor versus moderate SGN densities. In contrast, slopes of ECAP and EABR growth functions increased as a function of SGN density over a large range of SGN densities, so they could be used to distinguish moderate from poor SGN densities in the absence of IHCs. A few outliers were observed for all four measures, suggesting that there are variables affecting these functions that have not yet been assessed.

While additional data are needed, the results to date suggest that a combination of the tested measures could be useful in implanted ears for monitoring the efficacy and stability of treatments designed to preserve or restore the health of the cochlea. These measures could also allow identification of the healthiest stimulation sites when programming a patient's processor so that poor sites could be targeted for rehabilitation.

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W15: WHAT DOES SPECTRAL RIPPLE DISCRIMINATION AT HIGH RIPPLE DENSITIES MEAN IN COCHLEAR IMPLANT LISTENING EXPERIMENTS?

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Spectral ripple stimuli are increasingly being used as a psychophysical tool to assess spectral resolution in cochlear implant listeners. There are a variety of ways that these stimuli can be used, but some of the most common methods use a discrimination task that adapts on ripple density, with task difficulty increasing with increasing ripple density. Since cochlear implants have a fixed number of channels, there is an upper limit to the spectral detail that cochlear implants can transmit in their outputs. In this study, we investigate the electrode outputs of Nucleus cochlear implants with spectral ripple stimuli across a range of ripple densities. At low ripple densities, the analysis filters of typical cochlear implant processors can resolve individual ripples and this is observed as spatial modulations across the electrode outputs that match the spectral profile of the input signal. However, at higher ripple densities, individual ripples are not resolved and the spectral ripples of the input signal are no longer well represented across the electrode outputs and are either aliased or flattened. Based on electrode outputs, we observe that aliasing of high-density spectral ripples can result in output patterns mimicking lower spectral densities. This observation may explain why some subjects can discriminate spectral ripples at densities exceeding expectations based on number of channels. Our results suggest that researchers should approach the interpretation of experiments using spectral ripple stimuli at high spectral densities with caution, as these stimuli are often distorted or smeared by the signal processing of cochlear implant processors.

W16: MINIMUM AUDIBLE MOVEMENT ANGLE IN BILATERAL COCHLEAR IMPLANT USERS

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An important prerequisite to characterize the movement in relation to our own position is the ability to localize the sound sources. A common method to determine the precision of auditory localization is to measure the minimum audible angle (MAA), i.e., the just noticeable difference in azimuth at which a listener is able to differentiate between two stationary signals. Earlier studies showed that the performance of bilateral cochlear implant users (MAAs of 4 to 8° of arc) is close to that of normal hearing listeners (MAAs down to 1 to 4° of arc) for frontal sound incidence [Senn et al., 2005, Audiol Neurotol, 10, 324-352]. The present study investigates to what extent binaural cochlear implant users are also able to perceive movement of sound sources. The auditory resolution for moving sound sources was investigated by measuring the minimum audible movement angle (MAMA). For normal hearing listeners, the MAMAs for normal hearing listeners were found in the range from about 9 to about 20° of arc, depending on stimulus center frequency and velocity [Chandler and Grantham, 1992, J. Acoust. Soc. Am., 91, 1624-1636]. The present study investigates whether bilateral cochlear implant users can reach MAMAs comparable to those of normal hearing listeners. Moving sound sources were generated virtually on a semicircular arranged loudspeaker array consisting of 31 equidistant loudspeakers. Listeners were seated in the center of the semicircle with a diameter of 3.50 m. MAMAs were measured using continuous band limited noises. The influence of velocity and frequency content on auditory resolution was studied using stimuli with different center frequencies and different constant velocities. A stationary sound at 0° of arc (frontal incidence) was used as a reference. To evaluate individual performances due to overall differences in localization performance, MAAs were measured in addition to moving sounds for all listeners using identical stimuli. The results were compared to data from normal-hearing listeners using the same setup as for the cochlear implant user.

W17: EFFECTS OF STIMULATION MODE ON TEMPORAL PROCESSING BY COCHLEAR IMPLANT LISTENERS.

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Monopolar (MP), bipolar (BP) and tripolar (TP) stimulation modes may activate different sets of neurons and/or different sites of excitation (i.e., peripheral processes, central axons) in cochlear implant (CI) listeners. These different neural populations and/or different excitation sites may encode the temporal dynamics of the input signal differentially. The health of the surviving neural tissue at the site of excitation may further influence temporal coding for a particular stimulus. Here, we report on initial findings on pulse phase duration (PPD) sensitivity, temporal modulation coding, and recovery from forward masking using focused and/or monopolar stimulation modes.

Participants were adult and child users of Cochlear and/or Advanced Bionics implant systems. Stimuli were 300-ms long pulse trains presented at 1000 pps. PPD sensitivity was quantified in the form of slopes of threshold-vs.-PPD functions. Stimuli were presented in MP, BP or TP mode, and delivered to different electrode locations via custom research interfaces. Recovery functions were measured using different combinations of masker/probe stimulation modes. Maskers were loudness-balanced to a single reference (70% or 40% of the dynamic range). Temporal modulation detection was measured at carrier levels loudness-balanced to a single MP signal (40% of the dynamic range). Sinusoidal modulation was applied to PPD and/or pulse amplitude at rates of 100 Hz or 25 Hz.

Preliminary data indicate that PPD sensitivity is significantly greater with MP than with more focused modes. Recovery was significantly slower with more focused probes. In many cases, the recovery functions were also nonmonotonic with focused probes. Studies investigating the effects of masker stimulation level on recovery are ongoing. Temporal modulation sensitivity showed stronger across-electrode variation with BP mode than with MP mode. Ongoing studies are aimed at examining the inter-relationships between these different psychophysical measures of temporal acuity.

We speculate that the different patterns of results observed with focused stimulation modes reflect responses of smaller groups of neurons/localized stimulation of peripheral processes. It is possible that psychophysical thresholds obtained with MP stimulation tracks the responses of the most sensitive neurons. We therefore expect that in regions of excellent neural health, focused stimulation may yield sensitivity similar to that obtained with MP stimulation, whereas in regions of poor nerve survival, focused stimulation may result in poorer sensitivity. As PPD sensitivity has been shown to be correlated with nerve survival in animals (Prado-Guitierrez et al., 2006, Hear. Res. 215, pp. 47-55,), planned analyses of recovery and modulation sensitivity in relation to site-by-site PPD sensitivity may be informative about the functional state of surviving neurons. As focused stimulation is being actively discussed in the field, these results also have important implications for future speech processing strategies.

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W18: AUDITORY CONTEXT EFFECTS IN COCHLEAR-IMPLANT USERS

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Auditory context effects, such as loudness recalibration and spectral contrast enhancement, have been observed in normal auditory and speech perception. These effects may help "normalize" the incoming sound and produce perceptual invariance in the face of the variable acoustics produced by different rooms, talkers, and backgrounds. However, little is known about whether cochlear-implant (CI) users experience these effects. Discovering whether and how CI users experience auditory context effects should provide us with a better understanding of the underlying mechanisms, and may provide us with ways in which to improve CI processing to better approximate normal auditory perception.

In the present study, we examined the effects of a long-duration precursor on the perception of both loudness and (spectral or place) pitch of a short-duration target stimulus, which followed the precursor after a 50-ms gap. In separate experiments, CI users were asked to compare the loudness or pitch of the brief target with that of a comparison stimulus of the same duration that was presented after a 2-s gap. The assumption was that the precursor may affect the perception of the target but not the comparison stimulus, so that any differences in loudness or pitch would be due to the effect of the precursor on the target. The target stimulus was always presented to a middle electrode (electrode 8), and the position of the precursor was parametrically varied from electrode 2 through 14. The baseline comparison condition involved no precursor. When testing loudness, the current level of the precursor was set to 100% of the CI user's dynamic range (DR) and the target was to set to 70% DR. The comparison stimulus was varied between 40% and 100% DR within each block of trials. When testing pitch, the current of the comparison stimulus was divided in differing proportions between electrodes 7 and 9 to vary its perceived pitch between the values obtained for electrode 7 alone and electrode 9 alone. Similar conditions were tested in normal-hearing listeners using a noise-vocoder simulation of CI processing.

Preliminary data confirm effects associated with loudness recalibration and/or loudness enhancement in normal-hearing listeners, and suggest that some similar effects may be observed in at least some CI users. In contrast, no clear effects on spectral pitch were observed in the initial data. Further studies will examine the time courses of the context effects in both normal-hearing listeners and CI users.

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W19: SPREAD OF EXCITATION USING A NEW STIMULATION MODE, THE VIRTUAL TRIPOLE

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Spectral resolution is a limiting factor in performance with cochlear implants. Current shaping techniques have been used in attempts to improve spectral resolution. Current steering (e.g. virtual channels) has been used to increase the number of channels beyond the number of electrodes. Current focusing (e.g. partial tripolar stimulation) has been used to reduce the interaction across two channels in order to increase the independence of each channel and therefore improve spectral resolution.

We propose a new stimulation mode called a virtual tripole, which is a modification of a traditional tripole. The virtual tripolar is designed to reduce spread of excitation as well as allow for current steering. Tripolar stimulation consists of stimulation on one electrode combined with simultaneous out-of-phase stimulation on adjacent electrodes to reduce the spread of current. With a virtual tripole, each of the three electrodes in a tripole is replaced with a virtual channel, theoretically allowing for both current focusing and current steering.

Pilot data with four users of the CII or HiRes Advanced Bionics device has been collected. Spread of excitation was measured for monopolar, partial tripolar ($\sigma = 0.75$) and virtual tripolar ($\sigma = 0.75$ and 1) maskers located at electrode 9 using a forward masking technique. The probes were partial tripolar ($\sigma = 0.75$) pulse trains located at electrodes 6 through 12. Results suggest that a virtual tripolar masker provides a reduction in spread of excitation compared to the spread from a monopolar masker. Virtual tripolar ($\sigma = 1$) stimulation provides a narrower spread of excitation than partial virtual tripolar ($\sigma = 0.75$) stimulation. However, partial tripolar ($\sigma = 0.75$) stimulation using physical electrodes seems to produce a narrower spread of excitation than even a full virtual tripolar ($\sigma = 1$) stimulus. We will collect data from more subjects to have a more complete data set. Furthermore, we will evaluate pitch ranking for monopolar and tripolar virtual channels to determine their relative effectiveness at providing a continuous pitch percept. We expect to confirm that pitch changes monotonically for both monopolar virtual channel and virtual tripolar stimulation modes, as well as show a consistent reduction in spread of excitation with virtual tripolar stimulation compared to virtual monopolar stimulation.

W20: ELECTRODE SPANNING WITH PARTIAL TRIPOLAR STIMULATION MODE IN COCHLEAR IMPLANTS

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Recently, current steering has been introduced to the standard partial tripolar (pTP) stimulation mode in cochlear implants (CIs) by returning different proportions of current to the adjacent basal (α x σ x I) and apical [(1-α) x σ x I] flanking electrodes (Wu and Luo, 2013). With a fixed compensation coefficient σ, CI users perceived lower pitches with increasing steering coefficients α. Based on the results of a computational model (Goldwyn et al., 2010) and phantom electrodes (Saoji and Litvak, 2010), additional spectral channels may also be elicited in pTP mode by using non-adjacent rather than adjacent return electrodes on either side of the main electrode. This so-called electrode spanning or increased separation between the main and return electrodes may also be necessitated by a defective adjacent return electrode. In this case, current steering may be combined with electrode spanning to recover the channel in standard pTP mode.

These hypotheses were tested in five postlingually deafened adults using the HiRes-90K implant. The experimental stimuli were 300-ms, 1000-Hz, 226-μs/phase biphasic pulse trains delivered to the main electrode EL8 in various pTP modes. The highest σ allowing for full loudness growth within the compliance limit was used for each subject. The tonotopic pitch order was confirmed for standard pTP stimuli on main electrodes from EL6 to EL10, which were loudness balanced at the most comfortable level. In the first experiment, pTP stimuli with electrode spanning on either the apical [$pTP_{(6,8,9),\ \alpha=0.5}$], basal [$pTP_{(7,8,10),\ \alpha=0.5}$], or both sides [$pTP_{(6,8,10),\ \alpha=0.5}$] of EL8 were loudness balanced to the standard pTP stimuli on EL8 [$pTP_{(7,8,9),\ \alpha=0.5}$]. The pitches of the spanned pTP stimuli were compared to those of the equally loud standard pTP stimuli on EL6 to EL10. Results showed that in general, $pTP_{(6,8,9),\ \alpha=0.5}$ elicited a pitch percept between those of EL8 and EL9, while $pTP_{(7,8,10),\ \alpha=0.5}$ elicited a pitch percept between those of EL7 and EL8. Thus, each of them added a spectral channel between two adjacent electrodes. Surprisingly, $pTP_{(6,8,10),\ \alpha=0.5}$ were perceived as higher in pitch than $pTP_{(7,8,9),\ \alpha=0.5}$.

In the second experiment, current steering was applied to the spanned pTP stimuli, so that the center of excitation can be shifted back around the main electrode to recover the standard pTP channel in the case of a defective adjacent return electrode. $pTP_{(6,8,9),\ \alpha=0.25,\ 0.375,\ \ldots,\ 0.75}$ and $pTP_{(7,8,10),\ \alpha=0.25,\ 0.375,\ \ldots,\ 0.75}$ were compared to the equally loud $pTP_{(7,8,9),\ \alpha=0.5}$ in pitch. Results showed that $pTP_{(6,8,9),\ \alpha\approx0.7}$ and $pTP_{(7,8,10),\ \alpha\approx0.4}$ elicited similar pitch percepts as $pTP_{(7,8,9),\ \alpha=0.5}$, and thus recovered the standard pTP channel.

The pitch changes observed with pTP-mode electrode spanning and current steering were consistent with the shift of the centroid of computer-simulated excitation patterns or psychophysically-measured forward masking patterns. In summary, it is feasible to elicit additional spectral channels using spanned pTP mode, and to recover the standard pTP channel using both electrode spanning and current steering.

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W21: COMPARING MONOPOLAR WITH MULTIPOLAR STIMULATION MODES

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The monopolar electrode configuration is the most common stimulation mode used in today's cochlear implant (CI) systems. However, with better understanding and control of current interactions, it is possible to implement a stimulation mode that uses all electrodes to create a focussed electrical field. The goal of this project is to study the potential benefits of this multipolar stimulation mode and to compare it with the monopolar mode. Five patients have been implanted in Melbourne with a research device that is connected via a percutaneous plug to a benchtop stimulator providing 22 independent current sources. To date, three experiments have been performed.

In Experiment 1, 500-ms pulse trains were used in which two biphasic pulses were delivered on a single electrode with an inter-pulse delay varying from 1 to 5 ms within each 10-ms period. The subjects were asked to discriminate a standard stimulus with 1 ms delay to a stimulus with a different interpulse delay in a 3AFC procedure. The experiment was performed in both monopolar and multipolar modes. We define a reference delay as the delay at which subjects were able to discriminate the standard from the variable stimulus with an accuracy of 80%. On average, the reference delay was smaller by 0.35 ms in multipolar compared to monopolar mode. This suggests a better ability to discriminate small temporal differences in the multipolar mode.

In Experiment 2, stimuli also contained two pulses in each 10-ms period. The interpulse delay was set to the reference delay found in Exp. 1 for each subject. The second of each pair of pulses was delivered either on the same electrode as the first pulse or on a neighbouring electrode. It was hypothesised that channel interaction should decrease as the distance between the electrodes was increased, and that consequently it would be more difficult to discriminate different interpulse delays between electrodes that were further apart. If the multipolar mode produced more focused stimulation, such an effect should be stronger. However, no significant difference was found between the monopolar and the multipolar modes in this experiment.

In Experiment 3, the stimuli of Exp. 2 were presented with the electrode separation selected to result in chance performance, and this was confirmed experimentally. Then, the delay of the reference stimuli was set to 0 ms to test the hypothesis that there may be a mechanism that detects synchronies (rather than time delays) across cochlear places. The results showed scores above the chance level, supporting this hypothesis.

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W22: ASSESSING THE EFFECTS OF PULSE POLARITY ON SEQUENTIAL INTERACTIONS

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All modern cochlear implants (CIs) present pulses non-simultaneously in order to reduce charge interactions. However, the membrane of auditory nerve (AN) fibers needs time to return to its resting potential, causing the probability of firing in response to a pulse to be affected by previous pulses. This is particularly important for the high pulse rates used in contemporary CIs. Here we provide new evidence on the effect of pulse polarity on these interactions and demonstrate that, in a condition where loudness is assumed to be strongly affected by cathodic current, it can vary non-monotonically with level.

Seven users of the Cochlear Freedom device took part in this experiment. Detection thresholds and loudness balanced levels were measured for six electrical stimuli presented at 100 pps on a middle channel of the array. Two of the stimuli were anodic-1st (A) and cathodic-1st (C) single-pulse-per-period pulse trains. The remaining four stimuli were two-pulses-per-period pulse trains (TPPP) consisting of a biphasic pulse followed by another pulse of the same or of opposite polarity. All four combinations of pulse polarities were studied (“AA”, “CC”, “AC”, and “CA”). For TPPP, Thresholds were significantly lower when the adjacent phases of the two pulses had the same polarity (AC and CA) than when they had different polarities (AA and CC). About half of the subjects showed the threshold for AC to be lower than that for CA while the other half showed the opposite. This demonstrates for the first time that, at threshold, AN fibers are polarity sensitive and that a cathodic stimulus may sometimes be more effective than an anodic one. Contrary to threshold data, the stimulus that required the highest current to evoke a comfortably loud sensation was always AC, even for those who showed a lower threshold for AC than for CA. The level of AC had to be 2.9 dB larger than that of CA on average, consistent with previous findings showing that an anodic pulse is more effective than a cathodic pulse at comfortably loud level.

In an additional experiment, we asked the subjects to compare the loudness of 11 pulse trains of different levels (spanning the top ¾ of their dynamic range) both for AC and for CA. Several subjects showed a non-monotonic growth of loudness as a function of level for AC but not for CA. At some point in the dynamic range, loudness decreased with increases in level. This surprising finding may relate to a nerve conduction block produced by hyperpolarization of central processes and may have important implications for the selection of novel stimulation protocols in CIs.

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W23: EFFECT OF PULSE POLARITY IN BIPHASIC AND TRIPHASIC PULSES

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In cochlear implant (CI) patients the common stimulation pulse is a charge-balanced biphasic pulse. However, in animal experiments monophasic pulses result in higher efficiency as do asymmetric pulses and pseudo-monophasic trains [Wiler,1989, Hear Res, 39, 251-261], [Miller, 2001, Hear Res, 151, 79-94], [van Wieringen, 2005, Hear Res, 200, 73-86]. Pulse polarity has only small effects for biphasic stimuli, larger effects are expected for triphasic pulses. The influence of pulse polarity could depend on several factors: pulse shape, electrode position or species.

We tested 7 CI patients (MED-EL-Pulsar-CI100) with standard biphasic pulses (40µs phase duration) (reference) in comparison to triphasic pulses (same pulse length or phase duration). All leading-phase polarity combinations were investigated on two electrodes, 1 (most apical) and 6 (middle position). The task was to adjust the loudness (2-AFC, 2-down/1-up) of the triphasic pulses to the biphasic reference at a near-threshold level (threshold+25µA).

Our results showed that triphasic pulses are less efficient than biphasic pulses in accordance to [Coste & Pfingst, 1996, J Acoust Soc Am, 99, 3099-3108]. For triphasic pulses we found a strong and significant threshold difference when we switched pulse polarity, but only at the most apical electrode. At the middle electrode, changes were much smaller and not significant, as for biphasic pulses at both electrode positions.

We could replicate our findings with a multi-compartment Hodgkin-Huxley type neuron model in a homogeneous medium. Biphasic pulses were more efficient than triphasic. The position of the electrode relative to the neuron influenced the threshold difference between polarities as in the experiment. When the electrode was moved to the end of the neuron and beyond (position of apical electrode), the sign of the difference changed.

The model predicts that the apical electrode lies beyond the neurons, whereas the middle electrode is close to the end or even alongside the neurons.

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W24: ECAP-BASED MEASUREMENTS OF FORWARD MASKING FUNCTIONS

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Measuring psychophysical forward masking functions with cochlear implants typically involves presenting the Masker stimulus on a given stimulation site, followed closely by a Probe stimulus presented at either the same or at a different site. The Probe stimulation level is then varied adaptively to determine the point at which the Probe is just detectable. The Probe's detection threshold is therefore essentially the point where forward masking begins to be inadequate.
In principle, this threshold detection can also be measured objectively using the ECAP by considering the effect of inadequate masking. Adequate forward masking is a prerequisite for cancelling the stimulus artifact in ECAP measurements using the so-called Forward Masking Paradigm. By configuring these measurements such that a constant level Masker is followed by a Probe with varying stimulation levels, it is possible to objectively detect the point at which the amount of forward masking becomes inadequate, which in turn corresponds to the Probe's detection threshold.

A pilot study was therefore conducted to investigate the feasibility of such an approach, whereby forward masking functions were measured using the above method as well as using standard psychophysical procedures, and the results compared.

At the point of writing this abstract, complete results from a single subject have been collected and analyzed, and these are very encouraging, showing very good correspondence between objective and psychophysical data. Data collection is presently being conducted for 4 additional subjects and will be presented at the Conference.

Detailed analysis of the initial data also showed that the objective procedure can be simplified to further reduce the measurement time.

In comparison to the standard psychophysical approach, the ECAP based approach has the advantage of being faster, and also potentially easier to carry out as it requires much less subject interaction. Other implications of using such an objective approach will also be discussed.

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W25: EFFECT OF PULSE RATE ON THE DETECTION OF ELECTRICAL PULSE TRAINS BY USERS OF COCHLEAR AND AUDITORY BRAINSTEM IMPLANTS

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A series of experiments compared the effect of pulse rate on detection threshold for five cochlear implant (CI) and seven Auditory Brainstem Implant (ABI) patients implanted with Cochlear Corp. devices. A first experiment replicated a previously reported, surprising finding: for 400-ms pulse trains ABI users' thresholds dropped by about 2 dB as pulse rate was increased from 71 to 500 pps, but only by an average of 0.6 dB as rate was increased further to 3500 pps. This latter increase was much smaller than the 7.7 dB observed for CI users, and, for several ABI users, thresholds did not drop at all between 500 and 3500 pps despite the seven-fold increase in the number of pulses. We tested the explanation that, for ABI users, the neural response to high-rate pulse trains adapts markedly during the stimulus, thereby counteracting the effects of the higher rate. We observed a very similar pattern for shorter, 40-ms, pulse trains, thereby providing no evidence for the adaptation explanation. A second experiment investigated this further by measuring thresholds for 500-pps pulse trains consisting of 1,2,4,8, or 16, or 32 pulses, with, in one condition, each pulse being followed by six further 3500-pps pulses. The effect of these additional high-rate pulses remained much smaller for ABI users than for CI users, even for the comparison between a single pulse and a seven-pulse 3500-pps stimulus. Another possible reason for the small effect of pulse rate on ABI thresholds is that interactions between successive pulses, due to the finite time it takes neurons to return to their resting potential, increase at high rates. This would reduce the effectiveness of the stimulus when each pulse has the same leading polarity, because the second phase of each pulse has the opposite polarity to the first phase of the next. We therefore measured thresholds for three ABI users for fixed-polarity and alternating-polarity pulse trains. The reduction in threshold between 500 and 3500 pps for these three subjects was 1.2 dB for fixed-polarity and 2.2 dB for alternating-polarity stimuli, providing evidence for a minor role of sequential interactions. Additional hypotheses under investigation include the possibility that the effects of neural refractoriness are greater for ABIs than for CIs, thereby limiting any increase in neural response with increasing pulse rate.

W26: EFFECT OF MEDIAL-LATERAL ELECTRODE DISTANCE ON PSYCHOPHYSICAL AND OBJECTIVE MEASURES

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A negative correlation between cochlear implant recipients' levels of speech understanding and a measure of their average electrode-to-modiolus distances has recently been reported (Holden et al., Ear and Hearing, 2013). One hypothesis for why larger distances between the electrodes and the modiolus (i.e., higher "wrapping factors") correlate with poorer speech scores is that more distant electrodes may elicit broader neural excitation patterns that interact more with those from neighboring electrodes, thus negatively impacting the neural representation of speech. In our work, we examined the relationship between electrode-to-modiolus distance and sharpness of psychophysical tuning curves. These measures and analyses may give insight into the effect of distance on channel interaction. In addition, we examined the relationship between electrodes' impedances and distances from the modiolus.

Electrode positions were determined from analysis of high-resolution CT scans of the cochleae of CI users with percutaneous Contour Advance implants who had also undergone extensive psychophysical and objective measures. Psychophysical tuning curves were measured with a forward masking paradigm using focused multipolar stimuli. The probe was held fixed at a soft level while the masker level was adaptively varied to find masking threshold in a 3I-3AFC task. The masker position was changed to find the effect of masking at different spatial offsets from the probe.

The slopes of the tuning curves were an average of 4.3 dB/mm along the array for probe electrodes 1 mm from the modiolus. Electrodes at greater distances had shallower slopes indicating broader spread-of-excitation: with each medial-lateral millimeter of distance reducing the tuning curve slope by 2.8 dB/mm ($R^2 = 0.92$; $p = 0.002$; $F(1,4) = 47$; $N = 5$). These results suggest that greater electrode distances from the modiolus increase longitudinal interactions, at least with focused stimulation.

Finally, impedance at initial activation was found to be higher for electrodes closer to the modiolus, with impedance decreasing 2 k?/mm with increasing distance from the modiolar wall ($R^2 = 0.66$; $p = 0.01$; $F(1,6) = 11.6$; $N = 7$). The use of electrode impedance with other measures may provide a proxy measure for electrode distance in the absence of high-quality imaging data.

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W27: RATE DISCRIMINATION AT LOW PULSE RATES: COMPARISON BETWEEN NORMAL-HEARING AND COCHLEAR IMPLANT LISTENERS AND INFLUENCE OF INTRACOCHLEAR STIMULATION SITE

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Temporal pitch perception in cochlear implants (CIs) remains weaker than in normal hearing (NH), and is usually limited to rates below about 300 pulses per second (pps). Recent studies have suggested that stimulating the apical part of the cochlea may allow CI subjects to perceive increases in pitch up to higher rates than when stimulating other sites. The present study aims to better understand the limitations of the temporal pitch mechanism in CI users by focusing on rate discrimination at low pulse rates (ranging from 20 to about 100 pps).

This focus on low rates was motivated by data obtained in NH listeners showing that frequency difference limens (DLs) for bandpass filtered harmonic complexes were lower when the complex was filtered in a low than in a high frequency region (Krumbholz et al., 2000). Most importantly, this result was true even when the lowest harmonics present in the passband were clearly unresolved (rank > 12), suggesting that temporal cues are more salient when originating from the apex. However, given their stimuli all had a fixed bandwidth in Hz, it remained unclear whether the better performance observed in low frequency regions was due to the greater portion of neurons being excited or simply to the fact that it was lower.

Here, we present two experiments that measure and compare frequency DLs at four fundamental frequencies (F0s ranging from 20 to 104 Hz) in NH and in CI listeners. First, we replicate and extend the results of Krumbholz et al. by comparing DLs for harmonic complexes filtered in two frequency regions (lower cut-offs of 1200 and 3600 Hz, respectively) and for three different bandwidths (ranging from 250 to 6750 Hz). Second, we measure rate DLs for users of the Med-EL device at two different intracochlear sites. The two sites were chosen to be one of the three most apical and one of the three most basal channels of the device. DLs were measured for 500-ms stimuli in a 3I2AFC task.

For NH listeners, frequency DLs significantly improved as F0 increased from 20 to 104 Hz. For some subjects, performance was better in the low than in the high frequency region, even when the stimuli had the same bandwidth expressed in percentage of the filter's lower cut-off frequency. Although data collection with CI subjects is still being carried out, preliminary results suggest better performance for the apical than for the basal channel.

Krumbholz et al. (2000) JASA, 108(3): 1170-80

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W28: USING ELECTRODE DISCRIMINATION TO ENHANCE CLINICAL FITTING OF COCHLEAR IMPLANTS

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Cochlear implants (CI) are fitted on the premise that each electrode provides a distinct pitch percept at a discrete-spectral cochlear location. Thus loss of spiral ganglion cells may lead to neural dead regions or “holes” in hearing which could affect speech perception [Shannon et al., 2001, J Assoc Res Otolaryngol 2001; 3:185-199]. Simulation studies showed that redirecting information around a hole rather than dropping the information provides significant improvements in speech perception [Smith and Faulkner; 2006, J. Acoust. Soc. Am., 120:4019-4030]. Electrode discrimination (ED) using pitch ranking could be used to identify these “holes” and guide CI fitting.

Two studies were conducted to investigate the efficacy and effect of deactivating indiscriminable electrodes (IE). In the first study, a pitch ranking was used to determine ED and two research programs (RPs) were provided in a cross-over design to CI recipients with at least one IE. In the second study, the ability to rank the pitch of intermediate frequencies (IF) between a pair of adjacent active electrodes was evaluated in regions of (1) discriminable electrode-pairs and (2) indiscriminable electrode-pairs before and after deactivation. 35 adult cochlear implantees were recruited, 12 Med-EL, 14 Cochlear and 9 Advanced Bionics (AB). To assess the efficacy of the RPs speech perception was assessed following a one month trial. The Coordinate Response Measure (CRM) and BKB sentences in quiet at 70dBA and in noise at a signal-to-noise ratio of +10dB were used. Testing was conducted with the original clinical program (CP) and with each RP. RP options were: (1) allowing the stimulation-rate (SR) per channel to increase in AB & Med-El or increasing it manually in Cochlear, or (2) maintaining the original clinical SR.

Results of the best RP were compared to the best CP results and revealed: (1) that 20 of 25 participants demonstrated improvements (2) The t-tests conducted on BKB scores [transformed to rationalised arcsine units (RAU)] revealed significant improvements in BKB in quiet ($t = -3.47$, $df = 24$, $p < 0.01$) and in noise ($t = -2.90$, $df = 19$, $p < 0.01$). (3) The Wilcoxon signed ranks test was used to analyze CRM SRTs and showed significant improvement ($Z = -3.17$, $N = 15$, $p < 0.005$). (4) In contrast to participants demonstrating no benefit following the deactivation of IE, participants who received benefit showed better pitch ranking of IF in regions of discriminable electrodes compared to regions of IE and showed better pitch ranking of IF in regions of IE following deactivation compared to pre-deactivation. In summary, ED and IF pitch ranking test results can be used to improve fitting of CI.

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W29: COCHLEAR IMPLANT LISTENERS DEMONSTRATE ELEVATION PERCEPTION FOR SINGLE ELECTRODE STIMULI

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Normal-hearing (NH) listeners perceive different sound source elevations utilizing spectral cues (frequency-dependent peaks and notches) that are imposed by filtering from the pinna. An association between spectral cues and spatial elevation (i.e., a spatial map) are thought to be developed as one ages and gains experience with acoustic cues. In cochlear-implant (CI) listeners, the encoding of spectral cues is limited by poor spectral resolution, current spreads, lack of filtering from the pinna, and limited frequency range of the speech processor. It is thought that for these reasons, CI listeners cannot perceive changes in elevation [Majdak et al., Ear Hear., 32, 198-208, 2011]. However, these problems are inherent to the devices and the true ability of CI listeners to perceive changes in elevation remains unclear. To exploit the associations of elevation perception in CI listeners, we measured perceived intracranial elevation (i.e., “verticalization”) of simple delivered via direction stimulation of the speech processors. In NH listeners, sine tones that are higher in frequency produce the perception of higher elevation, independent of the sound source location. We hypothesized that CI listeners who have had some typical acoustic experience (late onset of deafness), when presented single electrode stimulation, would show similar patterns of elevation perception compared to NH listeners when presented sine tones. If elevation perception is developmental, CI listeners with an early onset of deafness who lack this experience would also lack these associations.

Fifteen CI listeners, eleven late onset of deafness and four early onset of deafness, were tested using constant-amplitude pulse trains at up to 13 electrodes, which were presented at comfortable levels. Pulse rates were 300 or 1000 pulses per second. Conditions with rate-roving were included (50% randomization of pulse rate) to confound assigning elevation to pitch cues. Listeners indicated the horizontal and vertical positions of an auditory image by clicking on the image of a face. Comparison data were also collected for six NH listeners with sine tone and Gaussian pulse trains using ear-insert headphones.

Eight of eleven CI listeners with a late onset of deafness generally perceived significantly higher verticalization at the basal-most electrodes and lower verticalization at the apical-most electrodes. Their responses were unaffected by rate-roving, implying that the verticalization was based on elevation perception rather than pitch cues. Two of four CI listeners with early onset of deafness showed a change in verticalization across the electrode array that was unaffected by rate-roving, thus indicating either some previous acoustic experience or innate association between verticalization and place of excitation. Further evidence that CI listeners perceived changes in elevation were that the verticalization trends and ranges in were similar between CI and NH listeners .These data highlight that elevation perception is accessible in CI listeners. Thus, there seems to be a possibility to provide vertical spatial cues to CI users with correct stimulus processing.

W30: SPECTRAL RIPPLE MODULATION DETECTION: EFFECT OF PHASE

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Psychophysical testing using spectral ripple stimuli has been widely used to assess spectral modulation detection thresholds (SMDT) for cochlear implant (CI) users. SMDTs have been shown to correlate with speech perception scores for CI users. Typically, the sinusoidal spectral ripple stimulus has been presented with randomized phase. Randomization of the phase has the effect of eliminating possibility of using cues from only one portion of the cochlea to perform the SMD task. However, it is unclear whether each ripple phase is equally well perceptible by CI listeners. The objective of this study is to assess whether the phase angle has a significant effect on spectral ripple thresholds in CI users.

The subjects were 10 cochlear implant users. SMDTs were measured for 1.0 and 0.5 cycles/octave. The SMDT task included four interleaved trials, each using a different starting phase for the spectral ripple (0, π/2, π, and 3π/2 radians). The procedure was a cued 2-interval forced-choice, with feedback, paradigm. The control stimulus was a flat spectrum noise and the other was a noise with sinusoidal modulations over 4 contiguous octaves (350-5500 Hz).

The spectral modulation detection thresholds varied within individuals when different beginning phases of the sinusoidal spectral stimulus were used for cochlear implant patients. More research is needed to establish the relationship and cause of differences between starting phase and SMDT. These data should be considered when using randomized beginning phase for SMD tasks.

W31: SPEECH PERCEPTION OF SECOND-GRADE CHILDREN WITH COCHLEAR IMPLANTS: COMPARISON TO TYPICAL CHILDREN

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Certain assumptions are so commonly accepted that their veracity is only rarely questioned. Among such assumptions are the ideas that specific properties, or cues, in the acoustic speech signal support the recognition of discrete phonemes for listeners, and once those phonemes are recovered from the signal they serve as the material for further language processing. These assumptions have a long history, rooted in the view of the nervous system as an input-output device. The data reported here, from children with CIs, raise questions about those basic assumptions.

Stimulus selection was based on earlier work demonstrating that children weight formant transitions more heavily in phonemic decisions than adults do, and increase their weighting of other cues as they mature. In this study two sets of edited natural stimuli were used: (1) stimuli varying vowel duration and 1st-formant offset transitions as cues to syllable-final stop voicing; and (2) stimuli varying fricative-noise spectra and 2nd/3rd formant onset transitions as cues to syllable-initial sibilant place. Predictions were that vowel duration would be heavily weighted by children with CIs because it is readily accessible through implants and fricative-noise spectra would be weighted very poorly because details of spectral shape are not readily available through implants. Formant transitions were expected to be weighted poorly, as well.
48 children with normal hearing (NH) and 51 with CIs participated in classic labeling experiments, and provided measures of speech recognition, auditory comprehension of language, expressive vocabulary, phonological awareness, and reading abilities. Results showed that children in both groups were similar in how they used vowel duration to label syllable-final stops, but children with CIs weighted formant transitions and fricative-noise spectra less than children with NH. For children with NH, only their weighting of fricative-noise spectra was correlated with language abilities, and in a positive direction. For children with CIs, only their weighting of formant transitions correlated with language abilities, and they were negatively related. These outcomes demonstrate that speech perception is quite different for children with CIs than for children with NH, and how that perception supports other language skills is not what would be predicted by the common view. While of interest to scientists seeking to improve cochlear prostheses, these findings also suggest that commonly held assumptions about how the nervous system handles sensory input for speech perception require some modification.

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W32: EFFECT OF MASKER FLUCTUATIONS ON LEXICAL SEGMENTATION IN COCHLEAR IMPLANT LISTENERS

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Normal-hearing listeners show better speech understanding in a fluctuating-amplitude masker than in a steady-amplitude masker, but most cochlear implant (CI) users show little or no masking release, even when masker fluctuations are timed to increase speech availability as in a previous study [B. J. Kwon et al. (2012). J. Acoust. Soc. Am., 131, 3111-3119]. The current study examined the hypothesis that the reduced or absent masking release in CI users might be due to factors in addition to energetic masking, namely disruption of their perception and use of metrical stress segmentation (MSS) cues. Eight CI subjects completed a sentence keyword identification task in a steady masker and a fluctuating masker with dips timed to increase speech availability. Lexical boundary errors in their verbal responses for each masker were categorized as consistent or inconsistent with the use of MSS cues. Results showed a positive correlation between the amount of masking release and the advantageous use of MSS cues in the fluctuating masker, supporting the hypothesis that CI users' reduced or absent masking release in fluctuating maskers might be partially due to difficulties in segmenting the portions of the speech signal available in the masker's dips.

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W33: APICAL VERSUS NON-APICAL ELECTRIC STIMULATION OF THE COCHLEA

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There are a number of variables that influence the wide range of postoperative speech perception results among cochlear implant recipients'. In theory, deeper electrode insertions could offer better low-frequency representation although at an increased risk of surgical trauma. Previous studies designed to assess the effect of insertion depth on performance have yielded conflicting results. These investigations have mostly utilized acute electrode deactivation strategies within subjects implanted with long arrays. This experimental design suffers from limited listening experience with the new maps and reduced channel numbers because of deactivation. Generalizations across subjects are further complicated by an inability to control for surgically-induced trauma. Similarly, comparisons across manufacturers have the limitations of differences in electrode design, number and spacing of stimulating electrodes, and signal coding strategy. The present study was conceived to assess the effect of apical cochlear stimulation while controlling for coding strategy, electrode number and the presence of surgical resistance and basal buckling.

Unilateral, conventional cochlear implant subjects were randomly assigned to receive either the standard (26.4 mm) or medium (20.9 mm) electrode array from Med-EL Corporation. Surgeons indicated the surgical approach, insertion depth, level of resistance experienced during insertion, and evidence of basal buckling. Postoperatively, speech perception was assessed with CUNY sentences, HINT sentences (quiet, SNR+10), and CNC words. Postoperative residual hearing and aided music perception abilities were also evaluated.

In the early activation period, patients with the longer array performed better than those with the shorter array. Preoperative or postoperative residual acoustic hearing did not explain these differences in performance. By one year of device experience, between group differences were less apparent. The influence of electrode length in the context of other medical/surgical and auditory variables will be discussed.

W34: THE BENEFIT OF HEAD ORIENTATION TO INTELLIGIBILITY OF SPEECH IN NOISE

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Speech reception thresholds (SRTs) in noise improve when the speech and noise sources are spatially separated. This spatial release from masking (SRM) is usually investigated in fixed-head situations. We studied free-head situations in audio and audio-visual conditions and compared normally hearing (NH) listeners' and bilateral cochlear implant (BCI) users' spontaneous or directed use of head orientation when attending to speech in noise.

A model was used to predict audio SRM. Predicted changes in SRM with head orientation away from the speech-facing orientation could yield a substantial benefit. We hypothesized that such benefit would motivate head rotation and that audio-visual presentation might discourage head rotation, possibly more so for BCI users who are more reliant on lip-reading.

In a first experiment, head orientation was tracked while NH or CI listeners attended to speech in noise in a sound-deadened room. Signal-to-noise ratio (SNR) was gradually diminished at a fixed overall level of 65 dB SPL. As SNR decreased, observed head tracks differed greatly between listeners, ranging from their remaining still, to gradually scanning for better speech intelligibility, to moving directly to the optimum predicted orientation. In the undirected task, listeners were found to be generally poor at optimally exploiting head orientation. The audio-visual condition reduced the amount of head rotation. In the directed task listeners made better use of head rotation.

In a second experiment, SRTs were acquired in the same room with fixed head orientations. The target speech was always maintained at 0 degrees, coinciding with the video of the speaker's face in the audio-visual condition. SRM was firstly measured with the masking noise at 90 deg or 180 deg with a 0 deg head bearing, secondly in the same spatial configurations but with a 30 deg head bearing that favored, where relevant, the listener's better ear. 30 deg was chosen as a head bearing that would improve SRM without impeding lip-reading. Predictions generated from head-related impulse responses acquired in the room with a KEMAR manikin were compared with the measured SRMs. This confirmed that a 30 deg head rotation yielded a maximum SRM of 8 and 6 dB for NH listeners and BCI users respectively. Audio-visual SRTs were found to be 3 dB below audio SRTs for NH listeners and 5-6 dB below audio SRTs for BCI users, illustrating their higher lip-reading ability.

The SNR attenuation reached in the first experiment reflected the SRM corresponding to the listeners' final head orientation. The two experiments combined demonstrate that BCI users could greatly benefit from training to optimize both their position and head orientation with respect to target speaker and noise source position in a social setting.

Research sponsored by Action on Hearing Loss

W35: CLINICAL FACTORS AFFECTING PERFORMANCE OF POSTLINGUISTICALLY DEAF ADULTS USING COCHLEAR IMPLANTS: A NEW CONCEPTUAL MODEL OVER TIME BASED ON 2251 PATIENTS

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Owing to space limitations, we were unable to include the names and affiliations of our 23 co-authors.

Data from 2251 patients implanted between 2003 and 2011 in 15 international centers were retrospectively collected to test the influence of multiple factors on cochlear implant (CI) speech performance in quiet and in noise for postlinguistically deaf adults. Results were compared to a similar study (Blamey et al, Audiology &Neurootology, 1, 293-306, 1996).

Speech scores in quiet and in noise were converted into percentile ranks to remove differences between centers. The influence of 15 pre-, per- and postoperative factors, such as the duration of moderate hearing loss (mHL), the amount of residual hearing, the use of hearing aids (HAs), the angle of insertion, the percentage of active electrodes, and the brand of device were tested. The usual factors: duration of profound HL (pHL), age, etiology, and duration of CI experience, that are already known to have an influence, were also included in the statistical analyses. The factors that were relevant in 1996 were still relevant in 2011, although their relative importance had changed. The significant new factors were: the pure tone average threshold of the better ear, the brand of device, the percentage of active electrodes, the use of HAs during the period of pHL, and the duration of mHL. Only 22% of the variance observed was explained by the factors tested.

A new model of predicted auditory performance with a CI was designed, showing a decrease of performance that started during the period of mHL, and became steeper during the period of pHL. The use of bilateral HAs slowed down the related central reorganization that is the likely cause of the decreased performance.

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W36: EFFECTS OF NUMBER OF ELECTRODES ON LISTENING EFFORT IN COCHLEAR IMPLANT USERS

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Cochlear-implant (CI) mediated hearing, due to device processing and a limited number of electrodes, lacks in both spectral resolution and temporal fine structure. Interpreting such a degraded auditory input requires increased cognitive processing. The processing demand of interpreting incoming sound or speech is referred to as listening effort. As long as the raised effort required to interpret the degraded speech does not exceed the limit of available cognitive resources, intelligibility may remain unchanged. However, increased listening effort is not without consequences, as it leaves less cognitive processing capacity available for concurrent tasks. This may lead to difficulties with selective or divided attention, verbal short-term memory, and ultimately verbal episodic long-term memory. Prolonged effortful listening can also result in fatigue, which in turn may lead to reduced overall performance, including speech understanding. These consequences illustrate the importance of optimizing listening effort for CI users.

In this study a dual-task paradigm is used to investigate effects of simulated or clinical device processing of the incoming speech signal on both intelligibility and listening effort. In the dual-task paradigm an intelligibility task and a visual response-time task are performed simultaneously. Effort associated with the intelligibility task is reflected in longer response times on the visual task, thus providing measures of intelligibility and effort respectively. Our previous study [Pals et al, JSHLR 2013] has shown, in normal-hearing listeners presented with CI-simulated speech of varying spectral resolution, that a dual-task paradigm can capture reductions in listening effort that are not reflected in speech intelligibility nor in subjective self-report.

The goal of the present study is to investigate the effects of spectral resolution on intelligibility and listening effort in real CI users. For each participant a number of experimental maps are created using fewer of the CI electrodes by assigning several frequency bands of the clinical processor to a single electrode. The results will yield insight into the dynamics of listening effort and intelligibility in CI users, and the dual-task as a measure of listening effort. This line of research can provide tools and strategies for assessing and optimizing listening effort for CI users.

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W37: IMPACT OF REVERBERATION ON SPEECH PERCEPTION WITH COCHLEAR IMPLANTS

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Introduction: Speech perception in patients fitted with cochlear implants (CI) decreases in reverberant environments compared to free field conditions. This holds even for small rooms, e.g. classrooms, with short reverberation times of a few hundred milliseconds. The results of previous studies demonstrate an exponential decrement in word recognition with increasing reverberation time (RT). Another detrimental factor is the direct-to-reverberant ratio (DRR), which describes the logarithmic ratio between direct and reverberant sound. The DRR decreases with increasing distance to the sound source. In the present study, the impact of DRR on speech perception in a group of CI users was examined and compared to a group of normal hearing subjects listening to CI simulations.

Method: Monaural room impulse responses of a typical classroom (RT = 0.75 s) and a medium sized church hall (RT = 4.1 s) were applied to generate reverberant stimuli with different RT characteristics. The diffuse reverb tail of the impulse responses was extracted and later in the experiment added to the direct signal (early reflections neglected). Sentence test stimuli were derived from the Oldenburg Sentence Test (OLSA) and recorded with a dual microphone hearing aid held by an artificial concha of a dummy head (B & K 4128). DRRs were in between -18 to +21 dB. Direct speech sound was presented via loudspeaker in frontal position (0°) whereas the reverberated signal was generated by 22 equally distributed surrounding plane waves created by wave fields (synthesis accomplished with 128 loudspeakers) to approximate a diffuse sound field. Speech reception threshold (SRT) depending on DRR was adaptively measured in quiet and in noise condition (10 dB SNR). 8 normal hearing (NH) subjects listened to a CI simulation realized with a 6-channel sine-vocoder (monaural headphone presentation). Furthermore, 6 CI-users were tested monaurally via direct connection to the auxiliary input of the CI-processor.

Results: Mean SRTs in classroom condition were about -18 dB DRR for both subject groups. With additional noise, a mean decrement of 2 dB was observed in the CI group. Mean SRTs in church condition in the NH group were -13.2 dB DRR (quiet) and -12.3 dB DRR (noise), CI group thresholds were -13.3 dB DRR (quiet) and -9.3 dB DRR (noise).

Conclusion: Over all, mean DRR results in quiet are comparable between real CI and CI-simulation group. However, an additional detrimental effect of noise was present in the CI group especially with long RT. DRR performance of the CI group without the presence of noise was better than expected. The perceptual problems with reverberation are not reflected in the present results. The DRR seems not to be the most important component to explain the perceptual difficulties reported by CI users in reverberant environments.

W38: SIMULATING THE BIMODAL EXCITATION SPREAD PRODUCED BY BIPOLAR STIMULATION: EFFECT ON SPEECH INTELLIGIBILITY

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Several studies comparing focused electrode configurations (bipolar or tripolar) with monopolar configuration have failed to find any benefits in terms of speech perception. One reason for this may relate to the fact that, in multipolar configurations, each electrode acts both as an active electrode and as a return electrode for different channels. A given electrode may therefore excite the same neural population with information extracted from different spectral bands. This effect should be biggest in bipolar stimulation where, for each channel, there may be two stimulation peaks of equal magnitude near the two stimulated electrodes. This study specifically investigates the perceptual effects of this bimodal excitation pattern using noise-vocoded CI simulations. We implemented a noise-band vocoder in which synthesis filters were designed to simulate the spread of activity in auditory neurons as predicted by a simple cochlear model. For each simulated channel, the synthesis filters were unimodal and bimodal for monopolar and bipolar modes, respectively.

Speech intelligibility was tested for nine normal hearing subjects using the French Matrix Test in the presence of a reversed speech masker. Experiment 1 tested three numbers of channels (4, 8, or 15), three simulated modes (monopolar "MP", bipolar "BP+1", control "CTRL"), and two target-to-masker ratios (+5 and +10dB). The CTRL mode had synthesis filters identical to the analysis filters. We found a significant interaction between stimulation mode and number of channels. Performance for MP and BP+1 clearly showed an asymptote at 8 channels whereas subjects continued to improve with CTRL up to 15 channels. Also, the improvement in scores as a function of number of channels was smaller for BP+1 than for MP or CTRL. This smaller improvement may be due to the increased overlap between channel for BP+1 when each simulated electrode starts to act both as an active and as a return electrode for different channels.

Experiment 2 investigated whether the spectral discontinuity introduced by the bimodal shape may be detrimental to speech intelligibility. BP+1 was re-tested and compared to two new synthesis filters (Continuous "CTN" and Asymmetric "AS"). CTN and AS were identical to BP+1 except that the hole between the two peaks was filled for CTN and one of the peak was removed for AS. CTN and BP+1 yielded equivalent performances suggesting that the spectral discontinuity introduced by the bimodal pattern may not be a problem. Furthermore, performance for AS was better than for BP+1 and CTN. This last observation encourages the use of asymmetric pulse shapes designed to attenuate the bimodal excitation produced by bipolar stimulation.

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W39: USE OF PSYCHOACOUSTIC MEASURES TO EVALUATE COCHLEAR IMPLANT CANDIDACY

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The primary audiological criterion for cochlear implant (CI) candidacy is best aided sentence recognition in the ear to be implanted of less than 50% with less than 60% recognition in the contralateral side. This requirement involves significant test time and clinical resources to ensure a “best-fit” aided condition. This protocol can also be problematic for patients who speak languages for which there are no available speech testing materials, for children, or for patients with multiple handicaps. The present study explores the possibility of using surrogate non-linguistic psychoacoustic measures without a hearing aid.

Eleven subjects with severe-to-profound hearing loss (20 ears) were recruited from patients visiting University of Washington Medical Center for CI evaluation. Spectral-ripple discrimination ability (13-dB modulation depth) and temporal modulation detection thresholds (TMDT) for 10 and 100 Hz modulation rates were evaluated with stimuli presented via insert earphones at comfortable loudness levels. Strong correlations were observed between spectral-ripple thresholds and both aided sentence recognition ($R = 0.77$, $p < 0.0001$) and unaided word recognition ($R = 0.74$, $p < 0.0001$). Weaker relationships were found between TMDT and speech understanding. In a subsequent subject, spectral-ripple scores accurately predicted a priori that the subject would not meet implantation criteria.

Results demonstrated that, for severe-to-profound hearing loss patients who might be CI candidates, unaided spectral ripple discrimination thresholds were well correlated with aided speech recognition. The unaided spectral-ripple discrimination test could be a promising tool for evaluating CI candidacy.

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W40: ROLE OF BINAURAL TEMPORAL FINE STRUCTURE AND ENVELOPE CUES FOR SPATIAL RELEASE FROM MASKING

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Spatially separating a speech target from interfering masker(s) generally improves target intelligibility; an effect known as spatial release from masking (SRM). This study assessed the contribution of binaural temporal fine structure and envelope cues to SRM for normal hearing listeners.

Target speech was presented from the front (0° azimuth) and speech maskers were either co-located or symmetrically separated from the target in azimuth (±15°, and ±90°) using KEMAR head-related transfer functions. The target and maskers were presented either as natural speech or as noise-vocoded speech. For the vocoded speech, intelligibility was conveyed only by the envelopes from 32, 16, 8 or 4 frequency bands.

Experiment 1 examined the effects of varying the number of frequency bands for the vocoder, and the degree of target-masker spatial separation, on SRM. For Experiment 1, vocoding was implemented prior to the introduction of spatial cues thereby preserving the spatial cues in both the speech envelope and the fine structure of the noise carriers. Results for Experiment 1 indicated that reducing the number of available spectral channels providing independent envelope cues systematically reduced the measured SRM. In Experiment 2, the vocoding was implemented post-spatialization to preserve the spatial cues mainly in the speech envelopes and not in the fine structure of the noise carriers. Preliminary results for Experiment 2 indicated that, for all frequency bands and spatial separations, SRM was significantly reduced when spatialization was done prior to vocoding.

Overall, the results highlight the role of binaural temporal fine structure cues for achieving SRM for normal hearing listeners. Potential implications for improving spatial hearing with bilateral cochlear-implant subjects will be discussed.

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W41: MEASURING SPECTRAL RESOLUTION FOR SPEECH SOUNDS: IMPLICATIONS FOR COCHLEAR IMPLANTS AND NEW CODING STRATEGIES

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Contrast in the auditory spectral (frequency) domain is essential to the accurate perception of speech, and is a primary limiting factor in performance with cochlear implants (CIs). In this study we introduce tests designed to directly evaluate listeners' resolution for spectral contrast in speech sounds. This has historically been difficult because of non-spectral acoustic redundancy in speech; listeners can rely on temporal cues or context to feign good spectral resolution. Direct tests of the spectral domain tend to be non-linguistic in nature (e.g. pitch ranking, spectral ripple discrimination), and are thus indirect evaluations of speech perception abilities. We offer a novel approach using modified natural speech sounds that are controlled tightly in specific acoustic domains, while retaining full spectral bandwidth and natural sound quality.

In this presentation we focus on speech sounds /b/ and /d/, which are notoriously confusable for CI listeners. These sounds were modified to vary orthogonally by spectral cues that were fine (second formant transition) or coarse (global spectral tilt) in scope; spectral resolution was measured as the reliance on the fine cue / suppression of reliance on the coarse cue, in accordance with previous studies of normal-hearing listeners. Responses from 12 CI listeners in a multiple-forced-choice identification task (including extra sounds used as fillers) were modeled using logistic regression, and model coefficients of each spectral cue were used as index of reliance (perceptual weight). Open-set word recognition scores were directly related to reliance on the formant transition cue ($r^2 = 0.42$, $p < 0.01$), and inversely related to reliance on the spectral tilt cue ($r^2 = 0.32$, $p < 0.01$). Overall word recognition is therefore at least partly driven by recovery of spectral cues, but is also affected by other factors besides spectral resolution.

Listeners with bilateral CIs were tested using their basic clinical maps and maps where odd- and even-index channels were delivered contralaterally (i.e. "interleaved" or "zippered" channels). Previous evaluations of this approach have been limited by the aforementioned confounds in basic speech tasks, so we applied our tests of spectral resolution to reveal potential benefit of interleaved channels, though to result from decreased electrode interaction. Listeners who showed poorest spectral resolution in their original maps benefited the most from channel interleaving, while listeners with already-adequate spectral resolution were minimally affected. Again, these patterns were not completely captured by open-set word recognition scores.

To the extent that improvements in CI speech coding are driven by innovations in spectral resolution, we propose that the approach used in this study can serve as a model assessment tool to evaluate new technology designed to improve spectral resolution, such as current steering and current focusing.

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W42: COMPARISON OF VOWEL CONFUSIONS FROM VOCODER SIMULATIONS IN NORMAL HEARING AND COCHLEAR IMPLANT LISTENERS

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Speech perception in cochlear implant users is impaired by reduced spectral resolution, which can be caused by regions of neuronal loss or electrodes located distant to auditory neurons. In the present study, simulations of such poor electrode-neuron interfaces were explored in normal hearing subjects, looking specifically at the impact on vowel recognition.

Two experiments were performed using a 15-channel noise band vocoder. In the first experiment the output of channels 6, 7, and 8 were modified to simulate a poor electrode-neuronal interface, encompassing the frequency range of the second formant of most vowels. Stimuli were presented with one of four manipulations to those channels; 1) Zero output ("Set to Zero", as in Shannon et al, 2002), simulating a loss of information, 2) Identical output ("Identical"), simulating complete channel interaction as might be seen with a large electrode-neuron distance, 3) Unmodulated noise output ("Unmodulated"), simulating a region where auditory neurons cannot respond to the spectral envelope information, 4) No manipulation ("All Channels") as a control condition. Ten normal-hearing listeners completed a 13-alternative, forced choice ASU Vowel identification task (Dorman et al., 1998) and contained no duration cues. Results indicated that vowel identification for the Unmodulated condition was significantly lower than all other manipulations, including the control condition.

In the second experiment, the impact of manipulating other vowel formants was explored by covering the frequencies of the first (1,2,3; apical), second (6,7,8; middle), and the third formant (11,12,13; basal). A second group of ten normal-hearing subjects were presented the vowels randomly manipulated by the Set to Zero or Unmodulated conditions and degraded through the apical, middle, or basal channels. An All Channels condition was again included as a control for a total of seven conditions. Consistent with previous studies, we found that the Set to Zero condition in the apical channels produced the largest decrease in vowel identification (e.g., Shannon (2002)). As predicted, vowel confusions for the apical manipulations occurred for vowels where F1 was degraded and vowel confusions for the middle manipulations occurred most when F2 was degraded. Manipulations of the third formant did not degrade performance as specifically as F1 and F2. These results suggest that errors in vowel recognition are predictable based on which channels contain the spectrally degraded signal.

A preliminary examination of vowel confusions obtained in cochlear implant listeners, for whom the locations of poor electrode-neuron interfaces were predicted based on elevated tripolar thresholds, demonstrated similar patterns of vowel confusion errors as those found in our normal-hearing simulations. A better understanding of the consequences of limited spectral resolution brings us closer to providing improved speech comprehension in cochlear implant listeners.

This work was supported by NIDCD DC012142.

W43: PSYCHOMETRIC FUNCTION OF RATE PITCH DISCRIMINATION DEPENDING ON JITTER APPLIED ON THE RATE OF ELECTRICAL PULSE PATTERN.

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The ability to discriminate rate pitch in cochlear implant users is poor compared to those with normal hearing. The strong entrainment of nerve fiber response to electrical stimulation is discussed as a possible cause since a collapse of stochastic neural firing patterns known from normal hearing (Johnson and Kiang, 1976) is observed. Restoring similar physiological stochastic firing may improve the just noticeable difference in rate (JNDR) (Morse and Evans, 1999; Rubinstein et al, 1999; Chen et al, 2005). A suggestion to overcome the entrainment is to jitter the pulse rate (Chen et al, 2005). However, up to now, no psychometric functions of rate pitch depending on the amount of pulse rate jitter have been determined. The expectation is that with increasing amount of pulse rate jitter in a certain temporal range JNDR decreases due to temporal information being hampered. Therefore, the present study addresses the measurement of JNDR depending on jittered electrical pulse rate in cochlear implant users.

Electrical pulse rate jitter was set either to either a Gaussian or uniform distribution. A psychoelectrical experiment in 11 subjects was conducted to assess JNDR in a three alternative, forced choice experiment. JNDR was recorded against the amount of jitter and corresponding psychometric functions were constructed. Average results showed decreasing JNDR with increasing amount of jitter independent of the applied jitter distribution. In conclusion, pulse rate jitter affects JNDR and therefore has to be considered in current coding strategies.

W44: CLINICAL ASSESSMENT OF SPECTRAL MODULATION DETECTION

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Spectral modulation detection (SMD), or spectral envelope perception, is a non-speech based hearing performance metric that provides a psychoacoustic estimate of spectral resolution. The electric SMD threshold of an implanted ear is highly correlated with speech recognition performance in the cochlear implant (CI) only condition1-6 and is thus a non-linguistic, psychoacoustic index of speech understanding performance. The research-based, psychoacoustic SMD task is expensive in terms of both software and time and complicated in setup and interpretation. Further the 30+ minutes required per ear to obtain a psychoacoustic SMD threshold for a single modulation rate renders it clinically prohibitive. We have invested significant effort to create a clinical version of this task that is simple to use (run from an audio CD or computer program and presented through the audiometer), time efficient (5-6 minutes per ear), accurate (correlated with the research task), convenient to score (pen and pencil or automatic via computer), and easy to interpret (results as 'percent correct' score). Our clinical task incorporates a 3-interval, forced choice procedure based on a modified method of constant stimuli. There are a fixed number of trials at each modulation depth (4 to 22 dB) and frequency (0.5 and 1.0 cyc/oct). Each trial is scored as correct or incorrect and spectral resolution is described as the overall percent correct score for the task (chance = 33%). Using a within-subjects, repeated-measures design with we describe data obtained from 20 cochlear implant recipients validating the abbreviated SMD task. The psychoacoustic task includes a 3-interval, 2-alternative, cued-choice adaptive procedure, where the modulation frequency (peaks/octave) is held constant and modulation depth (dB) is varied adaptively using a 3-down, 1-up stepping rule. The Pearson correlation coefficients for these data are 0.87 and 0.91 for the 0.5 and 1.0 cycle/oct, respectively, thus demonstrating the equivalence of the two measures. We have further compared the results of 60 experienced adult CI users for the clinical version of SMD and CNC word recognition. The results are consistent with a high correlation between outcomes (r=0.90) further demonstrating the efficacy of this clinical SMD measure as it has provided the same high correlation with speech recognition as shown for the research-proven, psychoacoustic metric1-6. These data represent the direct clinical translation of a research-proven task of spectral envelope perception into a clinically feasible format.

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W45: VALIDATION OF A CLINICAL ASSESSMENT OF SPECTRAL RIPPLE DISCRIMINATION ABILITY

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Spectral ripple discrimination (SRD) ability has been shown to correlate well with speech understanding in quiet and in noise for hearing impaired and for cochlear implant (CI) listeners [Henry et al., 2005 JASA; Won et al. 2007 JARO, 2011 JASA; Anderson et al. 2011 JASA]. Additionally, test results have shown a strong correlation with channel interaction [Jones et al. 2013 JASA]. The ripple test has furthermore provided a measure more sensitive than speech measures to CI processing modifications [Drennan et al. 2010, Hear Res], demonstrating differences acutely when CI users had no experience with the processing strategy. SRD has also shown stable results over a year of repeated testing [Drennan et al. 2011 CIAP] and spectral ripple tests have also shown promise for pre-operative clinical evaluations [Zhang et al. 2013 Ear Hear; Drennan et al. 2013 AAS; Moon et al. 2013 CIAP]. The SRD tests are however quite time consuming and not readily applicable for clinical use. A rapid “clinical ripple” test (about 8 minutes) has been developed and validated. Validation requires verifying test-retest reliability, verifying that the rapid measure is still clinically meaningful i.e. correlated with speech understanding, and verifying that the tests yields nearly the same outcome as the longer versions.

Twenty-eight CI recipients participated in the study. Each subject was tested on both the adaptive and the clinical versions of spectral ripple discrimination and with CNC word recognition in quiet. The adaptive version of spectral ripple employed a 2-up, 1-down procedure for determining spectral ripple discrimination threshold. The clinical ripple test used a method of constant stimuli, with trials for each of 12 fixed ripple densities occurring six times in random order. Results from the clinical ripple test (proportion correct) were then compared to ripple discrimination thresholds (in ripples per octave) from the adaptive test as well as with speech.

A strong correlation was found between the clinical ripple and adaptive ripple test results and a good correlation was found between both tests and word recognition. Excellent test-retest reliability was demonstrated with a test-retest correlation r = 0.9. The clinical ripple test is a valid, reliable and clinically meaningful non-speech measure of spectral resolution, optimized for use in a clinical setting.

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W46: EFFECT OF ELECTRODE IMPEDANCE ON ECAP SPREAD OF EXCITATION AND PITCH DISCRIMINATION WITH DUAL-ELECTRODE STIMULATION

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Several methods can be used to achieve intermediate pitch percepts in cochlear implants to theoretically improve spectral resolution. One of these methods, known as “dual-electrode” stimulation, electrically couples (shorts) two adjacent electrodes to effectively create a single “electrode” that should produce an electrical field that is different from that of either of the two contributing physical electrodes. Newer-generation Cochlear Corp. devices (24RE, CI512) have the capability for this type of dual-electrode stimulation. Because current takes the path of least resistance, impedance differences between the two coupled electrodes will likely affect the shape and location of the resulting electrical field. We expect that the lower-impedance electrode in each pair will dominate the response patterns obtained with dual-electrode stimulation. The shape of the electrical field affects neural spread-of-excitation (SOE) patterns, which can be measured using the electrically evoked compound action potential (ECAP). In this study, we hypothesize that large impedance differences between adjacent paired electrodes will result in ECAP SOE patterns for corresponding dual electrodes that approximate the pattern of the lower-impedance electrode in the pair. Furthermore, we hypothesize that pitch percepts between the dual electrode and the higher-impedance electrode of each pair will be more easily distinguished (than the dual electrode versus lower-impedance electrode) due to greater separation of SOE patterns.

In this study, ECAP SOE functions and psychophysical electrode-discrimination and pitch-ranking scores were obtained for pairs of adjacent electrodes with impedance differences of 3 kilohms or more, as well as for the corresponding dual electrode. Preliminary data suggest that the ECAP SOE pattern for the dual electrode more closely approximates that of the lower-impedance electrode in each pair. In several cases, pitch discrimination between the dual electrode and the higher-impedance electrode was better than discrimination between the dual electrode and the lower-impedance electrode in the pair. Results thus far suggest that dual-electrode stimulation may have limited benefit for improving spectral resolution for adjacent electrode pairs with large impedance differences.

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W47: DETERMINING THE BOUNDARY CONDITIONS FOR COCHLEAR IMPLANT MODELS

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Finite element models are useful in-silico tools for predicting the outcomes of scenarios that are commonly difficult to measure experimentally in-vivo. However, in order for the outcomes to be realistic, the input parameters for the models must also be accurate and reliable. These parameters include the material properties of the tissues, the boundary conditions, and the physical conditions applied. For the case of monopolar cochlear implant stimulation, finite element models of the cochlea have been produced to predict the current and voltage distribution within the cochlea and its surrounding tissues. In terms of boundary conditions, some previous models reported grounding the nerves and insulating the surrounding bone, whilst others have reported grounding the exterior limits of the model. Such assumption in boundary conditions can distort the current flow and lead to large differences in results.

This present study aims to determine accurate boundary conditions that can be used in localized cochlea models by retrieving the results from a whole head model. A three-dimensional human head model with twelve different tissues was created from anatomical images from the Visible Human Project. Electrical conductivity values were defined based on the values obtained from the literature. A purely stationary simulation of monopolar (MP1+2) stimulation was performed by injecting current into the cochlea with grounds set at the extracochlear return electrodes, which is considered much more realistic clinically.

The results from the simulation showed that the current does not only exit the cochlea via the nerve, but through several pathways that the previous models could not capture. Furthermore, there is clearly a non-zero, non-uniform voltage distribution in the tissues surrounding the cochlea. The distributions are plotted over boxes of different size and shape for use in a cochlea sized model for more accurate simulations.

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W48: AN INFORMATION THEORETIC FRAMEWORK FOR THE EVALUATION OF COCHLEAR IMPLANT CODING STRATEGIES.

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Over the last 40 years, different predictors of speech intelligibility have been used to account for speech signal distortion such as reverberation, additive noise or speech vocoding. However, these indexes are mostly based on the acoustic signal and cannot be used as predictors of speech intelligibility for cochlear implant (CI) coding strategies. Furthermore, speech intelligibility index have the limitation of being applicable solely to speech and do not give estimation of coding strategy performances in other listening condition (e.g. music perception).
We present here an information theoretic method which allows for an estimation of the speech intelligibility in the context of CI coding strategies. Using classical spike train analysis techniques, the mutual information content between a neural response and a set of stimuli is estimated. The calculation of mutual information is based on a classification using a nearest neighbour clustering algorithm. The distances between simulated auditory nerve spike trains strategy are computed using a one-parameter bin-less metric. Finally, we demonstrate that our method is not limited to the calculation of speech intelligibility and can also be used in the context of musical stimuli.

W49: DIMENSIONAL ANALYSIS OF THE HUMAN COCHLEA

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The human cochlea shows considerable interindividual variability in size and morphology. For development of atraumatic CI electrodes details of variability of human anatomy are required with high precision. For this purpose, X-ray microtomography (µCT) was used. The obtained data were imaged and dimensions of the cochlea were determined.

From fresh frozen temporal bones, blocks of approximately 3, 5 x 3, 5 cm were cut out around the cochlea. In order to visualize the soft tissues in the cochlea, the cochlear fluid was gently removed. The bones were scanned using a Skyscan 1173 machine (40-130kV source, 5Mp, 12 bit CCD sensor), resulting in images with pixel size varying from 8 -17 µm. Materialise MIMICS software was used to segment out the scala tympani (ST) and Rosenthal's canal from the obtained µCT images. A 3-D model of the segmented area was generated. Cross sectional images were taken perpendicular to the centerline of the ST. The 2-D images were exported to Matlab and analyzed by a custom-made software.

Comparison of different ST showed a large variability in the cross sectional diameter (CSD), vertical layout, and height of the ST. The CSD of the ST reaches a maximum within the first 2 mm and then overall decreases towards the apex. Relative standard deviation of the CSD is between 10 and 17%. Larger height values were measured at the center of the ST, compare to modialar and lateral region of the scala. At the lateral region, the height decreases significantly at the beginning of the 2nd turn.

In the vertical layout Z-dips were observed occasionally between 5-10 mm from the RW, and vertical jumps between 12-15 mm. The Rosenthal's canal (RC) extended up till 560 to 650 degrees. We found a correlation between the length of the RC and the length of the cochlea duct.

We segmented the ST and measured the internal dimensions using micro-computed tomography. We observed a large dimensional variability between different ST. These differences could have large implications in the design approach of CI arrays especially towards their capability to preserve residual hearing during insertion of the CI electrode array.

W50: A NEW GRAPHICAL METHOD FOR SHOWING THE ELECTRICALLY STIMULATED HUMAN COCHLEA.

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We present here a new, more intuitive, graphical representation of the electrically stimulated human cochlea. Cochlear implant researchers benefit from being able to visualize the position of the electrode array relative to potentially activated neural structures. For instance, picturing different insertion depths helps predict perceived pitch when stimulating individual electrodes. Researchers also benefit from a realistic representation of the distribution of the cochlea's neural structures. That is, it is important to appreciate the relatively shorter length of Rosenthal's canal (with resulting skewed distribution of spiral ganglion cells (SGCs)) compared to the full extent of the organ of Corti with its associated hair cells and peripheral processes (1.5 to 1.75 turns versus 2.5 turns, respectively). As compared to the cytocochleogram, our method is a more topographical representation of the coiled scala tympani which more accurately shows the nonuniform distribution of SGCs and displays a scaled representative number of the more than 30,000 hair cells, peripheral processes, and SGCs in the normal ear. In addition to helping visualize what an electrode is probably stimulating, this method lets a researcher predict potential "cross turn" stimulation and also provides for a graphical comparison of percentage nerve survival reported in temporal bone studies.

W51: OPTICAL STIMULATION OF THE COCHLEA - SINGLE CELL MEASUREMENTS ON HEK-HCLC CELLS AND SPIRAL GANGLION NEURONS

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It has been shown that inner ear cells can be stimulated in vivo by laser pulses resulting in cochlear potentials corresponding to auditory evoked signals [1]. Optical stimulation can be very site specific and hence because of the tonotopy of the cochlea very frequency specific, overcoming the limitations of conventional hearing aids as well as of electrical cochlea implants. To investigate the basic effects and mechanisms of optoacoustic stimulation, single cell measurements were performed.

Isolated cells of the cochlea, particularly spiral ganglion neurons, as well as transfected HEK-hClC-4 cells as model cells were stimulated with 5 ns laser pulses. Their electrophysiological reactions on different laser parameters such as pulse energy and wavelength were detected by means of the whole cell patch clamp technique.

The irradiated cells show a significant change in the measured membrane current, depending on wavelength and pulse energy of the laser light as well as the depolarization of the cell. These reactions are clearly elicited by the laser beam and can be observed at voltage clamp measurements as current spikes on a timescale of around 100 µs. The wavelength dependence of the cell reaction is in good agreement with the wavelength dependence of the absorption coefficient of water. Current clamp experiments on spiral ganglion neurons demonstrate similar reactions of their membrane potentials to the laser irradiation.

The results show cell reactions due to physiologically relevant optical stimulation in different cell types. Further studies will have to clarify to which extent the reaction is induced by thermal or optoacoustic stimulation or even the combination of both mechanisms.

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W52: INFLUENCE OF "DEAD" REGIONS ON THE TONOTOPIC REPRESENTATION OF SPEECH

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In the field of audiological diagnostics the identification of "dead" regions is crucial to obtain a better fitting of hearing aids. According to Moore and Glasberg 1997 [1], a "dead" region is defined as a location along the cochlea, in which the inner hair cells and/or the nerve cells are degenerated. For electrical hearing, this definition is reduced to the complete loss of spiral ganglion neurons (SGNs).

We have evaluated the influence of "dead" regions with our model of electrical hearing (Nicoletti et al., 2011 [2]) with statistically distributed SGNs. In our approach, the three-dimensional distribution of the surviving SGNs in Rosenthal's canal was randomized. From this distribution follows that each SGN is located at a defined distance from the stimulation electrode and therefore exhibits an individual threshold. This procedure explains several important observations: 1) a large dynamic range for electrical stimulation requires a large surviving SGN population (and vice versa), 2) the firing rates of individual SGNs are very heterogeneous along the cochlea and 3) the probability for the formation of "dead" regions increases with decreasing SGN number. Only for more than about 5,000 SGNs, the excitation patterns along the cochlea are sufficiently homogeneous to ensure that the relevant spectral components for good representation of speech are preserved. For lower numbers of SGNs, this representation is more and more deteriorated by dead regions.

This finding is consistent with observations from Blamey 1997 [3] and Khan et al. 2005 [4], who found the tendency, that a minimal number of surviving SGNs (around 5,000) is required for CI users to reach good word recognition scores. In conclusion, our model shows that for the performance of a CI user not only the absolute number of surviving SGNs is essential, but also how they are distributed along the cochlea and in Rosenthal's canal. With a number of less than 5,000 SGNs, the spectral coding worsens due to the formation of dead regions which in turn mask relevant spectral signal components.

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W53: A PHENOMENOLOGICAL MODEL OF THE AUDITORY NERVE FIBER RESPONSE TO MONOPHASIC AND BIPHASIC ELECTRICAL STIMULATION

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Models of the auditory nerve have the potential to aid the development of cochlear implant stimulation strategies. While biophysical models behave similarly to the auditory nerve in response to electrical stimulation, simulation with such models is computationally very demanding and inconvenient for studying population coding. Here we present a computationally inexpensive phenomenological model, based on the leaky integrate-and-fire model, with a novel extension to improve the realism of the response to a range of monophasic and biphasic stimuli.

Our model extends the leaky integrate-and-fire model by introducing a stochastic delay between threshold crossing and spike generation. An impending spike can be cancelled if sufficient anodic stimulation occurs during the stochastic delay. In response to monophasic stimuli, our model qualitatively reproduces the dependence of mean latency on stimulus intensity, while quantitatively reproducing both the dependence of jitter on stimulus intensity, and the input-output function at rest and during the relative refractory period.

In cathodic-anodic biphasic stimulation of the auditory nerve fiber, the leading (excitatory) phase of the stimulus injects charge, which is then removed by the second (inhibitory) phase. In this way, the biphasic threshold level is elevated relative to that of monophasic stimuli. This elevation in threshold can be diminished by introducing a gap between the two phases of the stimulus (the interphase gap). However, the threshold of a standard leaky integrate-and-fire model is unaffected by the interphase gap. This is because any occurring spike is evoked during the first phase of the stimulus, before the second phase can have an inhibiting effect. The stochastic delay of our model allows spike generation to occur during the second phase, and introduces the possibility of spike cancellation due to anodic current. With this extension, we have been able to reproduce the dependence of threshold on interphase gap, while simultaneously introducing realistic latency and jitter to the spike timings.

The main contribution of the model lies in its ability to simultaneously reproduce responses to a wide range of pulse shapes, including monophasic pulses and biphasic pulses of varying interphase gap. We intend to use the model to simulate populations of auditory nerve fibers under stimulation by cochlear implants.

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W54: MODEL-BASED EVALUATION OF SPEECH PROCESSING STRATEGIES IN A PITCH DISCRIMINATION TASK

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Cochlear implant (CI) recipients' auditory performance is affected by factors such as number of electrodes, electrode current spread, number of surviving auditory nerve fibers (ANFs), and the speech processing strategy implemented in the implant. Investigating the effects of any of the above factors by observing patients' performance is very expensive and time-consuming. This study proposes a model of auditory perception to predict CI users' performance in different tasks using different implant-related parameters, such as strategy, number of electrodes, rate of stimulation, etc., or CI recipients' individual parameters, such as the number of surviving auditory nerve fibers and the spatial spread of the stimulation current.

The input to the model is the simulated activity of the auditory nerve. The ANF responses are calculated either using an acoustic model, which simulates normal hearing, or a model of electric hearing, representing hearing with a CI. The model is a two-layer feed-forward neural network. The first layer has 200 neurons and receives the output firing rates of 200 simulated ANFs averaged over 100 ms. The second layer has a number of neurons equal to the number of classes to be recognized. Back-propagation of error is used as the training method.
The neural network was trained on classifying the pitch of synthesized vowels. 192 tokens from 16 types of vowels were created using a Klatt synthesizer, with fundamental frequencies spaced at half-octaves (viz. G2, C#3, G3, C#4). Normal-hearing subjects achieve near-perfect scores (100%) on this task, while CI users score about 60% correct. The vowels were divided into training set (129 tokens), validation set (15 tokens), and testing set (48 tokens). The model was trained using the acoustic model input and, as expected, reached a classification performance of 100% on the test set. The model was then re-trained using the electric model and the ACE strategy as input. The pitch recognition performance dropped to 63% on the implant test data, consistent with clinical results. Finally, to further evaluate the system, input from an experimental CI sound processing strategy that improved spectral acuity, was applied. It was observed that the model performance increased to 77%, suggesting that the new strategy may provide stronger spectral cues for pitch, compared to ACE. Also, the neural network reached its asymptotic level of performance with less training iterations with the new strategy than with ACE. This suggests that pitch perception may be learned faster by CI recipients using the new strategy.

This work demonstrates the possible benefits of the neural network model of auditory perception in the process of developing new stimulation strategies. Future work will examine the use of spiking neural networks in order to incorporate temporal aspects of the auditory nerve response.

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W55: MODELING AND EVALUATION OF LARGE SPIRAL GANGLION POPULATION RESPONSES

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Based on our large population model of the electrically excited auditory nerve we have developed various evaluation methods which provide quantitative comparisons on the quality of coding of different cochlear implant coding strategies. For our evaluations we usually calculate the spike trains of all surviving spiral ganglion neurons (SGNs) and we use a complex but realistic scenario: speech in noise. We evaluate the quality of speech coding with an automatic speech recognition (ASR) system. With temporal windowing (25 ms Hamming window) we evaluate the rate-place code of the neuronal spike trains. Our results show that all investigated coding strategies (HDCIS, FSP, FS4) provide high-quality speech coding. Recognitions scores were comparable to Mel-frequency cepstral coefficients (MFCC), the “gold standard” used in ASR systems. Also fine structure coding strategies (FSP, FS4) provide considerably more information in the time domain than HDCIS. If and how temporal fine-structure cues support speech understanding in noisy condition in the auditory pathway is still unclear. With yet another evaluation system we evaluate how well the location of a sound source is coded in auditory spike trains. For this case we apply a model of sound localization based on the correlation of spike trains from the left- and right ear. Also in this evaluation we could show that the higher temporal precision in the fine-structure channels provides just sufficient information for an at least coarse sound localization.

In summing up, our model and our evaluation strategies based on spiking auditory nerve responses and provide quantitative comparisons for the optimization of coding strategies. As the model of electric excitation of the auditory nerve also includes effects such as channel crosstalk, refraction and neuronal adaptation its predictive power goes far beyond pure analysis of the output patterns of implants, which is how contemporary coding strategies were developed. Where the final evaluation always of new strategies have to be done with CI users, this framework provides important answers to the question of the quality of coding available to the brain. Given the long development cycles including design, fabrication, approval, implantation, and finally extensive measurements in a large group of CI users to yield statistically significant results, the benefit of this approach cannot be overestimated.

W56: CORRELATION AND TEMPORAL PROPERTIES OF THE AUDITORY NERVE

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Infrared Neural Stimulation (INS) works by the local heating of tissue. A balance must be achieved between delivery and removal of heat, because an excess of heat delivered to the tissue results in damage. For INS, it has been shown that rates up to 200 Hz can be safely delivered. Here we explore a mechanism, by which stimuli can be applied while maintaining high frequency fine structure and minimizing the amount of heat delivered to tissues. We propose a Hidden Markov Model (HMM) can be used to achieve this task. Modeling of the spontaneous activity in auditory nerve fibers failed if an exponential (Poisson) or gamma distribution of the inter-spike time-intervals was assumed. Nevertheless, the distribution of the inter-spike time-intervals may still be described by a stochastic process. Spontaneous and driven activity of single pigeon auditory nerve fibers was tested for a renewal process. The experiments falsified mutually statistically independent lengths of time intervals between successive action potentials. The question for this study was if the generation of action potentials on the auditory nerve may be described by a doubly embedded stochastic model, the Hidden Markov Model (HMM), and whether the model parameters change for various inputs at different levels.

The analysis is an extension of the work by Richter et al., 1996. Spontaneous activity and responses to sinusoidal and white noise acoustic stimuli, which were recorded from single pigeon auditory nerve fibers, were used to estimate the transmission and emission matrix of the HMM. A newly generated sequence of events (surrogate data) was generated and the autocorrelation functions of the original data and the surrogate data were compared. Autocorrelation functions of the original data and surrogate data were not significantly different. Although action potential repetition rate was below 100 Hz fine structure of more than 1 kHz could be extracted from the spike trains. This fine structure was maintained in the surrogate data despite the fact that little correlation existed between the occurrence of the action potentials (recorded data) and the surrogate data. Fine structure of responses could be modeled with low event repetition rates.

The HMM provides a method to create sequences of events that contain similar high frequency structure as the original sequence of action potentials. The HMM might be used as a spike generator for auditory prostheses to stimulate the auditory system at low pulse repetition rates maintaining high frequency fine structure. This is particular attractive for infrared neural stimulation.

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W57: MYOGRAPHIC RECORDING OF THE ELECTRICALLY ELICITED STAPEDIUS REFLEX

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It is well-known that the electrically elicited stapedius reflex threshold (ESRT) correlates very high with the maximum comfortable level (C- or M-Level) and thus, is a viable tool to set upper stimulation limits in the fitting of cochlear implant subjects. Therefore, in many cochlear implant programs, ESRT is obtained intraoperatively by observing the contraction of the muscle. Postoperative evaluations of the muscle response by impedance bridge is very time consuming and in many cases not possible due to middle ear disturbances in the contralateral ear. The aim of this study was to investigate the feasibility of a myographic recording with an intramuscular micro needle electrode.

Different recording electrodes were developed to fit the placing requirements and penetrate the stapedius muscle. For eliciting the reflex a dual biphasic rectangular stimuli was used. Recordings were performed with a Nicolet Quest recording unit.

With this setting, up to date in more than 10 subjects tested ESRT could be obtained. When compared to the visually determined thresholds and later M-level, differences were not statistically significant.

This study demonstrates that myographic signals can reliably be recorded by an electrode from the stapedius muscle. However, more work is needed to optimize signal strength by recording electrode enhancement and stimulus optimization.

A further step could be to integrate this measurement method into a cochlear implant device. Then, ESRT-like ECAP measurements today - could be obtained postoperatively at any given time and would be an excellent parameter for fitting as well as long term observation of the thresholds of cochlear implant patient.

W58: ELECTROCOCHLEOGRAPHY ELICITED BY AN ELECTRICAL STIMULATION ON THE PROMONTORY

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Electrical stimulation of the cochlea can elicit an electrical potential travelling in the auditory pathway. In particular, in some previous studies, this electrical evoked potential can be recorded by a recording setup of EABR, and using at the same time trans tympanic electrical stimulation. For this reason, this type of measure is also called transtympanic EABR (TT-EABR). Using the electrical stimulation to the cochlea, and recording evoked potential for a 10 ms window after stimulus, it is possible to see if the electrical potential travels from the cochlea in the auditory pathway, up to the brainstem. The obtained response, evaluating by the amplitude/latency of the waves and generated in different points of the auditory pathway, can be used to decide if proceed or not with cochlear implantation, since with this method it can test the integrity of the auditory pathway, essential to get benefit of the use of cochlear implant. Such testing may be especially needed in case when it is not clear if cochlear implant candidate has functioning neural structures.

The goal of the study was to develop such testing system using the cochlear implant fitting system of MED-EL (Innsbruck, Austria). In this study, an electrical stimulation to the promontory is applied and an Electrocochleography is recorded. The evaluations of the results are given by the presence or absence of an electrical response in the cochlea within 6 ms from the applied stimuli.

A transtympanic electrode is used to stimulate electrically and it is placed through the tympanic membrane on the promontory of the cochlea, close to the round window, under general anesthesia. The reference electrode is placed on the ipsilateral temporal bone. The stimulation is generated by using the Detector Box from MED-EL. The Detector Box it is connected to the PC using the DIB II (Diagnostic Interface Box). The electrical stimulation is configured using the EABR task of the Maestro software (programming software for MED-EL audio processors). For one single channel of the Detector Box a certain level of current (in cu) can be specified. The full current range is divided in steps and they are applied from the lowest up to the highest level in order to obtain the Input/Output function. Other parameters can be configured. The parameter includes number of pulses, stimulation rate, pulse characteristics or trigger setting. For the recording of TT-EABR, an evoked potential recording device connected to the PC is used. A detailed set-up will be described.

To this date, we were able to record TT-EABR in 4 patients. Based on obtained responses it was decided whether the patient was implanted with cochlear implant or not. The initial data suggest that this tool may be very useful in difficult cases when it is not clear if patient benefited from cochlear implant or not. A further work is needed.

W59: AUDITORY EVOKED POTENTIALS WITH DIRECT ACOUSTIC COCHLEAR STIMULATION

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Cochlear's Direct Acoustic Cochlear Implant (DACI) Codacs® Investigational Device provides a direct stimulation of cochlear fluid through a stapes piston coupled to an actuator. The device could be indicated in patients with severe to profound mixed hearing loss. Little is known on objective assessment of an accurate coupling to the inner ear. A phase Ib clinical study has been performed in our center.

Methods: development of a test protocol for the measurement of auditory brainstem responses (ABR) and auditory steady state responses (ASSRs), demonstrating the coupling to the inner ear. Through bench tests, using an oscilloscope, some pitfalls in acoustic stimulation with DACI have been unraveled. Aided ABR and ASSR were measured with a single channel EEG set-up using RBA interface after complete implantation.

Results: Collected data show that artifacts complicate interpretation of responses in case of acoustic stimulation. By using a stimulus-controlled digital interface both ABR and amplitude-modulated noise ASSR have been successfully recorded.

Conclusion: preliminary results show that aided ABR and ASSR, demonstrating coupling to the inner ear, are feasible in bench tests and in Codacs subjects.

Support: FWO Flanders, IWT

W60: EFFECT OF TRIPHASIC AND BIPHASIC PULSE SHAPES ON THE SPREAD OF EXCITATION

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Clinically, a cochlear implant typically stimulates the auditory nerve with symmetrical biphasic monopolar pulses. The resulting broad spread of excitation induces interaction between the channels. A reduction in spread of excitation could improve the auditory performance of cochlear implanted patients in difficult listening situations. Similarly to a pseudomonophasic pulse, triphasic stimulation presumably emphasizes one of the two phases of electrical stimulation while maintaining a balanced charge. We are investigating the effect of biphasic and asymmetrical triphasic electrical stimulation on the spread of excitation of monopolar and phantom electrode configurations. Spreads of excitation are measured with Electrically Evoked Compound Action Potentials (ECAPs) using a forward-masked paradigm. We hypothesize that the use of asymmetrical triphasic pulses pulse would reduce the spread of excitation from pulse shapes that provide different spreads of current from each phase (such as phantom stimulation) but not when similar spreads of current are provided by each phase (such as in monopolar mode).

Preliminary data suggests that using asymmetrical triphasic pulses reduce the spread of excitation in both monopolar and phantom stimulation mode relative to biphasic stimulation. The peak of excitation is located near the recording electrodes in all the conditions. These preliminary results suggest that the use of asymmetrical pulses could reduce the spread of excitation and therefore potentially be used to improve auditory performance for cochlear implant patients.

W61: OF NOISE-SUPPRESSED SPEECH BY COCHLEAR IMPLANT USERS

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Many studies have reported that noise-suppression algorithms could significantly improve the intelligibility of noisy speech for cochlear implant (CI) users. While a number of intelligibility indices [e.g., the normalization covariance measure (NCM)] are available for reliably predicting speech intelligibility by normal-hearing (NH) listeners, little is known how well they predict the intelligibility of noisy and noise-suppressed speech by CI users. Access to such intelligibility measures is of great significance as they can be used to facilitate developing new noise-suppression algorithms for cochlear implants.

The present study assessed the performance of existing intelligibility and quality measures in predicting the intelligibility of noise-suppressed speech for Mandarin-speaking CI patients. Those measures were evaluated with intelligibility scores obtained by seven Mandarin-speaking CI listeners in 30 conditions involving noisy and noise-suppressed sentences corrupted by three maskers (i.e., speech-shaped noise, car and babble maskers) at different signal-to-noise ratios of 0, 5 and 10 dB and processed by four different noise-suppression algorithms. Correlation analysis was performed between the mean (across seven subjects) intelligibility scores obtained in each of the testing conditions and the corresponding mean intelligibility/quality measure values obtained across the 15 sentences in each condition.

Relatively higher correlation was obtained when using the envelope-based intelligibility index (i.e., NCM) than using the coherence-based speech intelligibility index (CSII), i.e., r=0.81 vs. 0.77. This may be partially attributed to the fact that present CI speech processors primarily deliver temporal envelope cues and little fine-structure information for CI speech perception. Additionally, some quality measures, e.g., the Perceptual Evaluation of Speech Quality (PESQ) measure, were also highly correlated with the intelligibility scores, i.e., r=0.82. This finding likely suggests that a measure that predicts reliably speech distortion (and overall quality) might also be suitable for assessing speech intelligibility by cochlear implant listeners.

Overall, the outcomes of the present study share several common findings with those obtained with wideband noisy and noise-suppressed speech by NH listeners. Specifically, the majority of the measures that were found to predict reliably the intelligibility of wideband noise-suppressed speech by NH listeners were also the ones found in the present study to predict well the intelligibility of noise-suppressed speech by CI users.

THURSDAY POSTER ABSTRACTS

R1: PULSE-SPREADING HARMONIC COMPLEXES: AN ADVANTAGEOUS CARRIER FOR SIMULATING COCHLEAR IMPLANTS.

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Vocoders are commonly used to simulate cochlear implants (CIs). In the vocoding process, envelopes extracted from various audio channels are imposed on new carriers, typically sine waves or Gaussian noises. Both of these carriers have limitations. A sine-wave carrier will stimulate a much narrower population of auditory neurons than that believed to be recruited by CIs and may additionally introduce audible sidebands. A noise carrier may overcome these limitations, but will contain intrinsic modulations that can distort the speech envelope and deteriorate speech intelligibility in a way absent for CI users. Finally, both carriers ignore the pulsatile character of the signals used in contemporary CIs.

Here, we present a novel carrier termed “pulse-spreading harmonic complex” (PSHC) that may overcome the limitations of both sine-wave and Gaussian-noise carriers. PSHCs are created by summing harmonically-related components with a specific phase relationship that allows us to generate acoustic pulse trains of various rates whilst keeping the fundamental frequency and, thus, the long-term spectrum constant. Simulations using a gammatone filterbank show that for a given characteristic frequency, there is an optimal pulse rate at which the intrinsic modulations of the stimulus after cochlear filtering are lowest. The amount of intrinsic modulations measured at the output of the auditory model is comparable across channels when using optimal pulse rates and relatively independent of the stimulus bandwidth as long as it is larger than the auditory filter bandwidth.

In a first step towards PSHC vocoding, this contribution focuses on the intrinsic modulations of various carriers and on how they impair the ability of normal-hearing (NH) listeners to detect an imposed modulation. Temporal modulation transfer functions (TMTFs) were measured in 7 listeners using a sine wave carrier as well as narrow (200 Hz) and broad (3200 Hz) bands of PSHC; of Gaussian noise; and of low-noise noise, all centered at 4 kHz.

Results show that TMTFs with a broadband PSHC carrier come closest to those obtained with a sine-wave carrier. This suggests that after cochlear filtering broadband PSHC contains the least intrinsic modulations apart from sine waves, making it a promising carrier for simulating CIs with NH listeners. Preliminary results on the intelligibility of PSHC vocoded speech and on how it compares to the intelligibility of sine-wave and Gaussian-noise vocoded speech will also be presented.

Work funded by the ANR (Project DAIMA)

R2: AMPLITUDE MODULATION DETECTION AND SPEECH RECOGNITION IN LATE IMPLANTED PRELINGUALLY DEAFENED COCHLEAR IMPLANT USERS

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Most cochlear implant (CI) users perform relatively well in ideal listening situations, i.e. speech understanding in quiet. However, late implanted prelingually deafened patients continue to struggle with speech recognition. This could be improved by matching the representation of relevant speech cues with the abilities of the patient's limited auditory nervous system. This study was conducted to evaluate the temporal processing abilities by means of temporal modulation transfer functions (TMTFs), and to determine whether these are related to speech recognition performance, in particular in late implanted prelingually deafened CI users.

Amplitude modulation detection thresholds (AMDTs) at 7 modulation frequencies including 5, 10, 50, 75, 100, 150 and 200 Hz, were obtained to assess the temporal processing of 8 prelingually deafened CI users and 18 postlingually deafened CI users. Phoneme recognition (CNC) scores (% correct at 65 dB SPL on an open-set Dutch monosyllabic word test) were gathered for all CI users. Scores on two additional speech tests for prelingually deafened CI users were obtained: (1) a monosyllable-trochee-spondee (MTS) test (% scores on word recognition and categorization of the word stress patterns administered at 65 dB SPL of a 12-item closed-set Dutch word test), and (2) a speech tracking test (the number of correctly repeated words per minute was scored) specifically designed for this population.

Results showed that prelingually deafened CI users tended to have a lower sensitivity to amplitude modulations and a higher TMTF attenuation rate than postlingually deafened CI users. None of the prelingually deafened CI users were able to detect modulations at 150 and 200 Hz. Also, 5 of the 8 prelingually deafened CI users were unable to recognize phonemes on the CNC test precluding further analysis. Word and stress pattern scores on the MTS test and speech tracking scores correlate significantly with the AMDTs at 5, 10, 50, 75 and 100 Hz modulation frequencies and with the TMTF attenuation rate. In postlingually deafened CI users CNC scores correlate significantly with 100 and 150 Hz modulation frequency and with the TMTF attenuation rate.

Prelingually deafened CI users are less able to detect amplitude modulations than postlingually deafened CI users. It appears that for prelingually deafened CI users, temporal processing abilities occur on a wide range of modulation frequencies (5-100 Hz) for speech recognition, suggesting that a speech coding strategy that focuses on temporal cues in the range of 5-100 Hz, might be beneficial for the speech understanding of prelingually deafened CI users. For the postlingually deafened CI users, phoneme recognition scores are mainly related to the temporal processing abilities on high modulation frequencies (100-150 Hz), indicating that focus on these frequencies might benefit speech recognition in this group.

R3: PERIODICITY AND APERIODICITY IN THE PERCEPTION OF SPEECH IN BOTH STEADY-STATE AND FLUCTUATING MASKERS

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Users of cochlear implants (CIs) experience great difficulties in speech perception in all types of background noise, and show little benefit from fluctuations in the masker. One popular (if partial) explanation for these difficulties focuses on a possible role of temporal fine structure (TFS) and the fact that CI listeners are relatively insensitive to TFS information. There is also a controversy about the extent to which TFS has a special role in allowing a release from masking in fluctuating noises, or whether its contribution is just as important for steady noises.

In order to address these questions, we are measuring speech reception thresholds in a series of related experiments. Both periodic and aperiodic target sentences are presented in both periodic and aperiodic nonspeech maskers, which can be steady-state or modulated in amplitude. The periodic maskers are based on fundamental frequency contours extracted from speech recordings of connected discourse (interpolated through periods of silence and voicelessness) and so mimic important aspects of real competing speech. Three types of listener are being investigated: CI users, listeners with normal hearing (NH) and NH listeners in CI simulations. Of course, the representation of any periodicity in the sounds presented will be much degraded in all but the NH listeners. Through these studies we hope both to pinpoint the exact role of periodicity for NH listeners (in particular, whether it has a special role in glimpsing), and the extent to which loss of information about TFS is able to account for the poor performance of CI users in noise.

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R4: LITTLE RED RIDING HOOD WAS A COCHLEAR IMPLANT USER!!

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To track a particular speaker in a crowd, or to simply recognize someone they know (perhaps a sickly Grandma), normal-hearing (NH) listeners rely on the perception of the vocal characteristics of the speaker. Two properties seem to be particularly important: the glottal-pulse rate (GPR), which relates to pitch, and the vocal-tract length (VTL), which relates to the perceived size of the speaker. NH listeners are very sensitive to both properties and interpret GPR and VTL cues to identify speakers' gender, age, size or other features [e.g. Smith and Patterson, 2005, JASA]. As such, how could Little Red Riding Hood (LRRH) have been fooled by the Wolf's voice, as the VTL of the Wolf would have been much larger than that of Grandma – big enough to "swallow her whole." We performed a series of experiments that may shed light on these long-ago tragic events, as well as on perception of speaker characteristics by cochlear implant (CI) users.

Little is known about CI users' perception of vocal characteristics. Previous studies suggested that, for voice gender identification, CI users rely almost exclusively on GPR, while NH listeners also use VTL [Fu et al., 2004, JARO; 2005, JASA]. However, in these studies, GPR and VTL were not manipulated parametrically but estimated from recordings of actual male and female talkers. In our first experiment, we systematically manipulated GPR and VTL from a single speaker in order to eliminate all other potential cues. Voice gender identification was measured for the various GPR-VTL combinations in CI users and in NH subjects listening to non-vocoded or vocoded speech (eight-channel CI simulation). GPR was the dominant cue for CI users (real or simulated) for voice gender identification, with little contribution of VTL.

In the first experiment, it was unclear whether CI users cannot perceive VTL cues, or whether they cannot use the cues correctly to interpret the size or gender of a speaker. In our second experiment, we measured just-noticeable-differences for VTL and GPR in real CI users, and in NH subjects listening to non-vocoded or vocoded speech. Preliminary results suggest that the VTL differences typically found between males and females are not salient enough to be perceived through CIs.

If LRRH was NH, we would have to conclude that the Wolf was not as big as presumed, and did not swallow her and the Grandma "whole" (sadly preventing any happy-ending to happen). However, if LRRH was wearing a CI, the present results indicate that she would not have perceived the longer VTL of the Wolf, and would have only noticed "what a deep voice you have" (pitch) and finally "what a big mouth you have" (visual cue) right before she was eaten. Improving perception of VTL cues, maybe through cognitive training, may improve CI users' speaker identification (and could save them from being eaten by wolves).

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R5: TRADING FUNDAMENTAL FREQUENCY FOR DURATION CUES IN LEXICAL TONE RECOGNITION BY TAIWANESE CHILDREN WITH COCHLEAR IMPLANTS

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Lexical tones are an integral part of tonal languages. Fundamental frequency (F0) variations provide critical information for lexical tone recognition, while other acoustic cues (e.g., intensity or duration characteristics) may serve as secondary cues. Previous studies have shown that cochlear implants (CIs) do not transmit F0 cues well, resulting in deficits in melody recognition, speaker identification, intonation recognition, and lexical tone recognition by CI listeners. For Chinese-speaking children acquiring their native language with the help of CIs, the ability to identify and distinguish lexical tones is an important perceptual skill. We speculate that in the absence of a salient F0 cue, these children may rely on the secondary acoustic cue, namely, duration. In this study, we examine acoustic cue-trading between F0 and duration in tone identification. The single word "yanjing", was used as the stimulus. Spoken with a flat tone, the word means "eyes"; spoken with a falling tone, it means "eye glasses". The naturally produced word was resynthesized to generate 2, 8, and 6 steps of variations in the acoustic dimensions, i.e., F0 height, F0 contour, and duration ratio, respectively. Eleven children with normal hearing (NH) and 24 CI children (all 6 years of age and older) heard a single presentation of each stimulus and were asked to indicate whether they heard "eyes" or "eyeglasses". In addition, overall tone recognition accuracy was also measured using naturally produced monosyllabic words. Preliminary analyses using logistic regression indicate that in the yan-jing task, the NH children relied significantly more than CI children on F0 contour variation for lexical tone identification, while CI children relied significantly more on the duration cue than their NH counterparts. Overall tone recognition accuracy was near ceiling in the NH children, but as expected, was poorer and varied considerably among the CI children. These results are broadly consistent with previous findings on speech intonation recognition by CI and NH listeners, and again indicate the importance of redundancy in speech perception. Further analyses, including a comparison of psychophysical pitch discrimination measures in these children with their performance in tone recognition, and effects of age of implantation, chronological age, etc. are ongoing and will be presented as well.

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R6: EVALUATION OF A REAL-TIME IMPLEMENTATION OF AN EXPLICIT PITCH CODING STRATEGY FOR COCHLEAR IMPLANTS

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Pitch is the perceptual counterpart of the fundamental frequency (F0) of a sound and is important for the perception of melodies, intonation in speech and recognition of lexical tones in tonal languages. Several studies have demonstrated significantly worse pitch perception for cochlear implant (CI) users than for normal hearing listeners. For this reason, during the last 10 years our research group has developed a CI processing strategy called F0-mod (Laneau et al. 2006; Milczynski et al. 2009). In the F0-mod strategy all electrical stimulation channels are sinusoidally modulated in amplitude according to an F0-estimate.

F0-mod was perceptually evaluated in pitch and tonal-language speech perception experiments, and significant improvements over the standard clinical ACE strategy were found. However, an off-line MATLAB implementation of F0-mod was used, using sophisticated F0-estimation and voicing detection algorithms with manual fine-tuning, that cannot be implemented in a real-time system.

In the present work, we implemented F0-mod on a real-time (frame-based) system, and evaluated it physically and perceptually in similar conditions to those presented by Milczynski et al. (2009).

For the physical validation, the response of the real-time system to pure tones, complex tones, and speech tokens at different intensities was measured and compared with the existing off-line implementation using the following measures: average energy per channel, fundamental frequency and modulator signal as a function of time, and visual inspection of electrodograms.

For the perceptual validation, pitch and speech perception were measured in CI users. The electrical stimulation patterns were generated in real time and directly streamed to the subjects' implant using the xPC system provided by Cochlear Ltd. The following pitch experiments were conducted: (a) Pitch Ranking, where the subject had to indicate whether the reference or the test tone was higher in pitch; (b) Melodic Contour Identification, for a sequence of musical tones, where the subject had to indicate the visual counterpart of the perceived pitch contour; and (c) Familiar Melody Identification, for a selection of nursery rhymes, where the subject had to indicate the correct song title. The speech experiments carried out were: (a) three-phoneme word recognition in quiet; and (b) sentence recognition in noise.

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R7: RELEASE FROM MASKING OF SPEECH IN 8 HZ SAM NOISE IN THE ABSENCE OF TEMPORAL FINE STRUCTURE

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Cochlear implant users and normal listeners presented with vocoded speech in noise often fail to show the advantage from amplitude modulation of the masking noise that is seen with normal listeners and unprocessed speech. This has been attributed to the absence of temporal fine-structure information. However, the lack of such a dip-listening advantage may depend upon the nature of the task and the signal-to-noise ratio (SNR).

Low signal-to-noise ratios and strongly modulated noises might be expected to facilitate dip-listening, especially in combination with relatively predictable closed-set speech identification tasks. We have studied the perception of speech in steady-state and 100% modulated 8 Hz SAM speech-spectrum noise through an 8-channel noise vocoder with two closed-set tasks: the Tyler Matrix Sentences with four slots each with four alternative responses, and the identification of one of nine vowels in a carrier phrase. In a group of 10 normal listeners, 50% correct SRTs for unprocessed speech in unmodulated noise were around -12 dB, while SRTs in 8-channel noise-vocoded conditions were in the range of -3 to -6 dB. Both unprocessed and vocoded conditions showed similar degrees of masking release, of approximately 3dB, with SAM noise compared to unmodulated noise. A masking release with modulated noise compared to unmodulated noise in vocoded conditions was also found by Ihlefeld et al (2010, JASA, 128:870-880) with a Co-ordinate Response Measure task and 16 Hz square wave modulated noise.

The masking release seen here cannot depend upon temporal fine structure information, and an enhancement of such masking release could be one factor in the benefit often conferred by residual hearing in EAS or bimodal stimulation. Additional data from implant listeners and simulated bimodal listening conditions will be reported.

Supported by Action on Hearing Loss.

R8: RESULTS IN ADULT COCHLEAR IMPLANT USERS WITH ASYMMETRIC HEARING LOSS

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Traditionally, cochlear implant (CI) recipients have had moderate to profound hearing loss in both ears. Individuals with asymmetric hearing loss; that is, moderate to profound hearing loss in one ear and better hearing in the other ear, have not been implanted routinely because of their better hearing ear. Often these individuals have discontinued amplifying the poorer ear due to lack of benefit and wear amplification only in the better hearing ear. They are, therefore, unilateral listeners. This study examined performance and outcome variables in adults with asymmetric hearing loss who received a CI in the poorer ear to determine whether this treatment improved communication function.

Twenty adults with asymmetric hearing loss (one ear meeting CI candidacy criteria and the other ear with better hearing) were tested pre-implant and at 1, 3, and 6 months post-implant. Pre-implant, participants were tested with the better ear aided, the poorer ear aided (better ear plugged/muffed) and in their everyday listening condition (unilateral or bilateral hearing aids). Post-implant, participants were tested with the better ear aided, the CI only (better ear plugged/muffed) and bimodally. The test protocol was designed to incorporate measures and conditions that simulated real-life listening challenges (e.g., background noise, varied speaker locations and loudness levels, and multiple talkers). Test materials included CNC words, HINT sentences in noise, TIMIT sentences in quiet and noise, BKB-SIN, HINT sentences in the R-SpaceTM, and localization measures using a 15 loudspeaker arc. Additionally, a self-assessment of perceived communication function (SSQ) was completed at each interval.

Group mean results indicated significant open-set speech recognition in the implanted ear after 6 months for tests in quiet and noise. Furthermore, significant improvements in speech recognition and localization were seen when comparing the 6-month bimodal condition to the pre-implant, everyday listening condition. Mean SSQ ratings indicated improved perceived communication function at 6 months post-implant compared to pre-implant. Variables that affect outcomes for traditional CI users also affected outcomes for these non-traditional users (i.e., deafness duration and age at onset). Overall, results indicated fitting each ear with the appropriate sensory device provided improved bilateral hearing and communication function in adults with asymmetric hearing loss.

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R9: ACOUSTIC SPECTRO-TEMPORAL MODULATION DETECTION IN COCHLEAR IMPLANT USERS

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While there are many benefits of cochlear implants (CIs), including partial restoration of hearing loss, there are still many difficulties for CI recipients. One of these is that speech understanding with CIs is poorer and more variable than in normal hearing subjects. Speech contains combinations of spectral and temporal modulations that convey essential acoustic information for speech understanding in quiet and in noise. In NH subjects, degraded temporal (Drullman et al 1994), spectral (vanVeen & Houtgast 1985), or joint spectro-temporal cues (Elliot & Theunissen 2009) can adversely affect discrimination and intelligibility of speech. Spectro-temporal modulation sensitivity and its role in speech recognition have not been extensively studied in CI patients. Since CI patients receive partial or degraded spectro-temporal cues, it is desirable to identify what spectro-temporal cues are available and how they are utilized by CI patients. This study aims to 1) measure the joint spectro-temporal cues that are available to CI patients and subsequently 2) determine whether spectral and temporal cues are independently or jointly processed by CI patients.

We measured modulation detection thresholds monaurally using ripple sounds with purely spectral (0.5, 1, 2, 4, 8 cycles/octave), purely temporal (8, 16, 32, 64, 128, 256 Hz) and joint spectro-temporal modulations. Nine CI subjects were tested with stimuli delivered to the sound processor via the auxiliary port and nine control NH subjects were tested over ER-2 headphones. The temporal (tMTFs), spectral (sMTFs) and the spectro-temporal (stMTFs) modulation transfer functions were derived as a function of temporal modulation (Hz) and spectral modulation (cycles/octave). All MTFs exhibited low-pass functions for both dimensions. Both CI and NH subjects exhibited comparable sensitivity to purely temporal modulations up to 128 Hz. However, CI sensitivity to spectral modulations (upper limit 4 cycles/octave) was degraded compared to NH subjects (8-16 cycles/octave). CI subjects exhibited lower thresholds and thus improved sensitivity to spectro-temporal modulation over both spectral-only and temporal-only sensitivity. Surprisingly, while stMTFs of NH subjects where separable functions of temporal and spectral modulation, as previously reported (Chi &Shamma 1999), stMTFs of CI listeners where not separable. This indicates an interaction between spectral and temporal modulations that enhances detection of joint spectro-temporal cues not evident in NH subjects.

The results indicate that future CI processor designs can take advantage of the enhanced spectro-temporal modulation sensitivity in CI patients to potentially improve speech recognition and intelligibility. This study also suggests that spectro-temporal modulation detection can serve as a useful measure of the clinical sound processing strategies than purely temporal or spectral modulation detection measurements.

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R10: INFLUENCE OF EXPANDED VOICE PITCH CONTOURS ON THE INTELLIGIBILITY OF SPECTRALLY REDUCED SPEECH

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Variation in the fundamental frequency (F0) of the talker's voice is an important cue for speech recognition. When F0 variation is reduced, intelligibility declines, particularly in complex acoustic backgrounds. Cochlear implant (CI) users demonstrate reduced sensitivity to changes in F0, and show limited benefit from these cues in noise.

While some CI users can improve perception of voice pitch by combining electric stimulation with residual low-frequency hearing (electric-acoustic stimulation), their performance in noise typically falls short of that achieved by normal-hearing listeners. Previous research has shown that synthetically expanding word-level F0 variation can enhance its salience for users of electric stimulation. This suggests that naturally produced speech with expanded F0 variation may also provide a benefit for CI users.

The present study investigated the effects of voice pitch variation on sentence intelligibility in simulated electric-only and electric-acoustic stimulation. This necessitated the creation of a new speech corpus with specific acoustic parameters. The corpus consists of a closed set of 50 monosyllabic words, divided equally into five categories (name, verb, number, color, object). A male and a female talker recorded each word twice, once with a normal F0 contour, and once with an exaggerated contour. Each talker's voice pitch was carefully controlled, such that the mean F0 of each talker is consistent across the two contours, and the degree of F0 variation in each male contour is proportional to that in the corresponding female contour. Stimuli were edited to control for amplitude and duration differences. When combined, the words form syntactically correct but semantically unpredictable sentences containing well-defined normal or expanded pitch contours (e.g., Bob found eight gold bags).

Target sentences were mixed with single- or multi-talker maskers at several signal-to-noise ratios, then vocoded to simulate CI stimulation. The target-plus-masker mixture was low-pass filtered at 500 Hz and combined with the vocoded stimuli to simulate electric-acoustic hearing. Normal-hearing listeners were tested monaurally over headphones on all talker/contour combinations in simulated electric- and electric-acoustic stimulation. Sentence intelligibility scores indicate that performance trends higher with expanded F0 contours than with normal contours, but only for male target speech. This suggests that expanded F0 contours may benefit users of electric stimulation in complex acoustic backgrounds, although degree of benefit may depend on the target talker's mean F0.

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R11: TEMPORAL RESPONSE PROPERTIES OF THE AUDITORY NERVE IN CHILDREN WITH AUDITORY NEUROPATHY SPECTRUM DISORDER

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Cochlear implantation has been used to manage hearing deficits for children with auditory neuropathy spectrum disorder (ANSD) who do not benefit from hearing aids. It is still unknown how well neural synchronization can be restored at the level of the auditory nerve via electrically stimulation for individual patients with ANSD. This project aimed to 1) evaluate temporal response properties of the auditory nerve at different pulse rates in cochlear implanted children with ANSD; and 2) explore whether the observed temporal response properties correlate with speech perception performance in these patients.

Twenty two implanted children with ANSD ranging in age between 4.1 to 18.2 years participated in this study. All subjects were implanted with Cochlear Corp devices and had at least six months of listening experiences with their devices. All subjects had no known anatomical malformations or developmental delays. Temporal response properties of the auditory nerve were evaluated by measuring amplitudes of the electrically evoked compound action potentials (ECAPs) in response to individual pulses in a pulse train delivered at four different rates: 500, 900, 1800, and 2400 pps per channel. The pulse train was presented at the maximum comfortable level measured at 2400 pps. ECAPs were recorded for one apical, mid-array and basal electrode for each subject. Phonetically Balanced Kindergarten (PB-K) Words lists were presented at 65 dB SPL with monitored live voice to evaluate open-set speech perception abilities.

Results showed that ECAPs recorded from some subjects showed diminished amplitude with increasing stimulation rate, consistent with increased neural adaptation. However, ECAPs recorded in a subgroup of subjects showed increased amplitudes over the pulse train at all four stimulation rates. In addition, these subjects tended to have worse speech perception scores when compared to those who showed typical temporal response patterns.

These results suggest that some implanted children with ANSD have atypical temporal response patterns at the level of the auditory nerve. This, in turn, suggests incomplete restoration of neural synchronization via electrical stimulation in these patients.

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R12: EFFECT OF CONSONANT-VOWEL BOUNDARY TO SPEECH PERCEPTION IN COCHLEAR IMPLANTS

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Speech sounds are broadly classified into vowels and consonants. Consonants carry more information than vowels while reading as there are more consonants compared to vowels in English language. Despite being less in numbers, vowels contribute more to speech intelligibility than consonants. Previous research with normal hearing listeners showed that the vowel-only sentences (consonants replaced) led to a 2:1 intelligibility advantage over consonant-only sentences (vowels replaced) regardless of the type of segmental replacement. However, the contribution of consonants and vowels to speech perception in cochlear implant (CI) listeners is not well understood.

A segment replacement paradigm on TIMIT speech corpus was used to evaluate the effect of consonant-vowel boundary to speech perception in CI listeners. Two processing strategies were created to emphasize the duration of vowels and consonants by presenting different amounts (0%, 40%, 60%, 80%, and 90%) of consonants and vowels. The first strategy (FVXC) preserved full vowels (FV) and presented different amounts of consonant (FV+%C) by replacing the consonant centers with either silence or speech shaped noise. The second strategy (FCXV) preserved full consonants (FC) and presented different amounts of vowel (FC+%V) by replacing the vowel centers with either silence or speech shaped noise. Clean speech and interrupted speech (3 Hz interruption rate, 50% duty cycle) were also presented. A PDA based research platform was used to present the stimuli to CI listeners.

For FCXV presentation condition, noise replacement paradigm had higher speech intelligibility compared to silence for 0%, 40%, and 60% vowel presentation conditions. There was no difference in speech intelligibility for 80% and 90% vowel presentations. Moreover, there was no significant difference between the replacement paradigms for FVXC presentation condition. Also, the speech intelligibility was very low (< 10%) for 3Hz interruption suggesting the inability of the CI listeners to tolerate interruptions in continuous speech. Overall it could be concluded that the CI listeners have difficulty in fusing interrupted speech signals into one coherent speech stream and phonemic restoration in CI listeners occurs depending on the presented speech stimuli.

R13: IS WEAK PITCH FULLY RESPONSIBLE FOR LOW COMPREHENSION OF INTERRUPTED SPEECH WITH POOR SPECTRAL RESOLUTION?

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Cochlear implant (CI) users have great difficulty to understand interrupted speech. It has also been suggested that they may benefit less from the phonemic restoration that occurs in normal-hearing (NH) listeners when the interruptions are filled with noise [Başkent, 2012, JARO]. Although the restoration is considered to be based on top-down mechanisms, it is also thought that the saliency and continuity of primary acoustical cues also play a role. Two factors could therefore reduce the restoration in CI listeners. First, the poor spectral resolution could increase the ambiguity of the presented speech segments, offering a weaker basis to infer the missing segments. Second, the weak pitch delivered by the implant could affect the perceived continuity of the speech signal. We showed in a previous study [Clarke et al, 2013, ARO] that the continuity of vocal characteristics (including pitch) is not necessary to reconstruct interrupted speech. However, the presence of pitch in the speech signal could help make the distinction with the noisy, pitch-less interruptions, when spectral resolution is poor. Failing to discriminate the noise from the speech could lead CI users to erroneously interpret the noise as a speech segment. We thus expect restoration to fail when the pitch cue is completely removed from the speech signal.

To separate the effects of pitch presence and spectral resolution on restoration, we created unvoiced speech (no pitch cues but full spectral resolution), noise-band vocoded speech (no pitch cues) at different spectral resolutions and the same vocoded speech with additional pitch cues. These manipulations allowed us to investigate pitch presence and spectral resolution independently, whereas the weak pitch cue is usually linked to the poor spectral resolution in CIs. We measured speech intelligibility of NH participants with the synthesized voices in a phonemic restoration paradigm, i.e. with silent interruptions and with these interruptions filled with speech-shape noise.

Preliminary data show, as expected, an overall decrease of intelligibility as the spectral resolution is reduced. It seems, however, that the effect of pitch presence or absence does not follow exactly our expectations. It is possible that NH listeners rely on different cues given the situation in order to take full advantage of the top-down mechanisms involved in restoration. Our results will be discussed within the framework of interrupted speech to explain what we better understand about restoration, but also in the light of studies seeking to improve the pitch representation in CIs [e.g. Brown and Bacon, 2009, JASA].

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R14: PRESTO: PRELIMINARY FINDINGS WITH A NEW HIGH-VARIABILITY SENTENCE RECOGNITION TEST IN PATIENTS WITH COCHLEAR IMPLANTS

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Understanding speech in noise continues to be one of the most common problems facing patients with cochlear implants, and is a topic of much research. To perform well in everyday noisy environments, listeners must quickly adapt, switch attention, and adjust to multiple sources of variability in both the signal and listening environments. Sentence recognition tests in noise are useful for assessing speech recognition abilities because they require a combination of basic sensory/perceptual abilities as well as cognitive resources and processing operations. PRESTO (Perceptually Robust English Sentence Test Open-set) was developed at Indiana University as a sentence level test of speech perception that more closely approximates real-world speech communication with multiple talkers, engages more cognitive processing resources, and does not reach ceiling levels of performance in quiet, to allow for analysis of individual differences. PRESTO differs from other sentence recognition tests because target sentences differ in talker, gender, and regional dialect. Previous work in this lab has evaluated performance on PRESTO with normal hearing listeners, hearing-impaired listeners, and non-native speakers of English, demonstrating that individuals varied in the their ability to perform the task and that the variation in performance was related to differences in executive function, specifically, working memory capacity, selective attention, and control of inhibition. This new study extends the prior work to determine if this same relationship exists in a clinical population of post-lingually deafened adults with cochlear implants.

The goals of the current study were to compare PRESTO with two conventional clinical tests of sentence recognition (AZBio and HINT) in quiet and noise, and to explore the relationship between demographic, cognitive, and self-assessment measures with sentence recognition, to characterize the underlying information processing skills associated with speech perception in adverse conditions. In this study, post-lingually deafened adult CI participants were tested with two sentence lists (quiet and +10SNR multi-talker babble) from each sentence test (HINT, AZBio, and PRESTO). Participants also completed a working memory span test, the Speech and Spatial Qualities Questionnaire (MRC), the Behavior Rating Inventory of Executive Function (BRIEF-A), and the Word Fam test.

Preliminary results from N=8 patients demonstrate a significant main effect of sentence test materials (HINT>AZBIO>PRESTO) both in quiet and in noise. Scores on PRESTO were significantly poorer and displayed greater variability than HINT or AZBio scores. There was also a significant interaction between test and noise condition; PRESTO was more challenging in more difficult listening conditions. PRESTO scores were highly correlated with measures of auditory working memory capacity. Relationships between speech perception scores, executive function, and self-reported scores on the SSQ Questionnaire will be discussed. PRESTO is a perceptually robust test of speech perception and is a valuable addition to the “clinical toolbox” to assess speech perception skills of patients with hearing loss in quiet and multi-talker babble that may be useful for pre- and post-cochlear implantation evaluation.

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R15: EFFECT OF INHERENT AMPLITUDE FLUCTUATIONS IN STATIONARY NOISE ON SPEECH INTELLIGIBILITY IN COCHLEAR-IMPLANT USERS

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Understanding speech in noise remains a major challenge for cochlear-implant (CI) users. Even in normal-hearing (NH) listeners, it is unclear how much masking is due to the overall noise power (as is mostly assumed), and how much is due to inherent amplitude fluctuations of noise. Recent studies by Stone et al. (2011; 2012) have suggested that inherent noise fluctuations play a major role in limiting the intelligibility of speech. They used NH listeners and vocoder processing to dissociate masker amplitude from masker fluctuations. CI listeners provide a rare opportunity to study the dissociation of masker amplitude fluctuations from masker spectrum without further processing, due to the fact that CI processors generally extract the temporal envelope from within each frequency channel and discard the temporal fine structure.

In this study, we compared the effects on sentence intelligibility of three masker types. The first masker (PT) had no inherent fluctuations, and was produced by adding pure tones at frequencies corresponding to the center frequencies of each CI channel. The second masker (GN) was a broadband Gaussian noise. The third masker (MT) was comprised of pure tones, as with PT, but with each tone amplitude-modulated with the temporal envelope of Gaussian noise within that CI frequency channel. All maskers had the long-term spectral envelope of speech. We predicted that the GN and MT maskers should produce the same amount of masking in CI users, because they produce similar outputs from the CI channels. Earlier results from NH listeners suggested that the PT masker would produce much less masking, due to a lack of inherent amplitude fluctuations. Controls were NH listeners, who listened to either the unprocessed stimuli, as presented to the CI users, or to the same stimuli processed through a tone-excited envelope vocoder.

Results from the NH listeners in vocoded conditions showed clear detrimental effects of inherent fluctuations, consistent with earlier data. Also consistent with expectations, preliminary data from CI users showed no difference between the results using GN and MT maskers. However, in contrast to expectations, preliminary data from CI users showed no clear benefit of eliminating inherent fluctuations in the PT condition. Overall, it seems that CI users may not derive the same benefit from a lack of inherent masker fluctuations as do NH listeners.

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Stone MA, Fullgrabe C, Moore BCJ (2012) Notionally steady background noise acts primarily as a modulation masker of speech. J Acoust Soc Am 132:317-326.
Stone MA, Fullgrabe C, Mackinnon RC, Moore BCJ (2011) The importance for speech intelligibility of random fluctuations in "steady" background noise. J Acoust Soc Am 130:2874-2881.

R16: ASSESSING LISTENING EFFORT IN NORMAL-HEARING AND COCHLEAR IMPLANT RECIPIENTS THROUGH USE OF A DUAL-TASK PARADIGM

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Many individuals who use bilateral cochlear implants (CIs) or both CI and hearing aid (HA) report that less effort is required to understand speech when listening with both devices than with a single device, especially in noise. Traditional tests of speech recognition, however, provide no measure of the effort required to understand speech, and therefore may not fully capture all of the benefits provided by use of CIs or hearing aids (HAs). One approach to assess listening effort is through use of a dual-task paradigm. In this paradigm, the listener divides attention between a primary and secondary task. As the primary task becomes more difficult, fewer cognitive resources are available for the secondary task, generally resulting in poorer performance.

Here, we used a dual-task paradigm to assess listening effort in both normal-hearing (NH) listeners and recipients of CIs. In both cases the primary task was to repeat AzBio sentences, and the secondary task was to recall a string of 5 or 7 digits presented visually before a set of either 2 or 4 sentences. As controls, the primary and secondary tasks were each tested singly, and compared with results obtained in the dual-task paradigm. The NASA Task Load Indices for Effort, Mental Demand, and Performance were also completed in each test condition. For the NH listeners, the AzBio sentences were processed with an 8-channel CI simulation, and speech recognition was assessed in quiet and in background noise. CI recipients were tested with the AzBio sentences in noise (+10 dB SNR) in unilateral and bilateral / bimodal conditions.

Our results from the NH listeners indicate that performance on the primary sentence-recognition task was not affected significantly by the addition of a secondary task, suggesting that we can measure listening effort without compromising our ability to assess speech recognition. Moreover, performance on the digit-recall task decreased significantly between the single- and dual-task paradigms, suggesting that this measure is sensitive to changes in listening effort. There was a trend toward increased listening effort with decreasing SNR, although there was considerable intersubject variability. Finally, the NASA Task Load Indices for perceived effort, mental demand, and performance largely mirrored the behavioral data, albeit with considerable variability which may preclude its clinical use. We are currently collecting data from individuals using bilateral CIs or CI plus hearing aid to assess the utility of the dual-task paradigm for measuring listening effort in these groups of patients. These results will be reported.

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R17: EXPLORING MASKING RELEASE FOR DIFFERENT MASKER MODULATION RATES USING A NEAR-STEADY MASKER

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Stone et al. [(2012) J. Acoust.Soc Am. 132:317-326] showed that “modulation masking” rather than “energetic masking” was dominant when a “steady” noise was used as a masker of speech. They also showed that “masking release” (MR), the difference between the intelligibility of speech in a modulated and steady background, was small or absent when the background was composed of multiple widely-spaced sinusoids, and had minimal amplitude fluctuations. This was shown only using a modulator with an 8-Hz rate. Here, intelligibility was measured for a wide range of modulation rates. Following Stone et al., speech was filtered into 28 channels. To minimize peripheral interactions between channels, odd-numbered channels were presented to one ear and even to the other. Speech intelligibility was assessed in the presence of a sinusoidal masker centered on each band. The masker was steady or was 100% sinusoidally amplitude modulated at rates from 1 to 81 Hz. The target-to-masker ratio was adaptively varied to track 60% word intelligibility. Preliminary results confirm that MR does not occur for a masker with minimal amplitude fluctuations unless the modulated masker used for comparison fluctuates at rates less than 4 Hz

R18: A TWO-MICROPHONE SPEECH ENHANCEMENT ALGORITHM DEDICATED TO A BILATERAL COCHLEAR IMPLANT

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In quiet listening conditions most cochlear implant (CI) users can now achieve good word recognition scores regardless the used device. However, speech recognition scores are highly degraded in noisy environments. Furthermore, as mentioned by CI users, better and comfortable speech recognition in noisy environments would be considered as one of the most significant challenges. Now, to improve speech intelligibility in noisy environments, individuals with severe to profound hearing loss could be implanted with two cochlear implants, one in each ear. In fact, bilateral cochlear implantation provides patients the advantages of bilateral information. Bilateral hearing permits a better understanding of speech in quiet and also in noisy environments. However, under more challenging listening conditions, BCI recipients perform poorly, compared to normal-hearing listeners. Numerous speech enhancement algorithms, divided into single-microphone and multi-microphone methods, have been proposed over the years to improve speech recognition in noisy background conditions.

Coherence based methods have been successfully applied to dual-microphone noise reduction systems. These techniques showed good results when noise signals on two microphones were uncorrelated, but their performance decreased with correlated noises. It could be improved when the cross power spectral density (CPSD) of received noises is available. In this paper, an improved minimum tracking (IMT) technique for noise CPSD estimation was proposed. The performance of this technique was compared to two other noise CPSD estimators based on voice activity detection (VAD) and minimum tracking (MT) approaches.

Performance evaluation was performed at four signal-to-noise ratios (SNR) and two interfering noise source configurations (single and three interfering noise sources). Subjective listening tests were carried out with 50 normal hearing listeners using a specific bilateral cochlear implant simulator and utilizing the French Lafon database corrupted by additional babble noise. Objective measures based on PESQ scores indicated a significant improvement in speech quality when the IMT approach was considered. Subjective recognition tests revealed that CPSS algorithm can yield substantial benefits in speech intelligibility for Bilateral CI in simulation when the IMT approach for noise CPSD estimation is considered, especially in the presence of only one interfering noise source.

R19: A PITCH SYNCHRONOUS FRAMEWORK FOR REPRESENTING, RECOVERING, AND MODIFYING TEMPORAL STRUCTURE IN SPEECH

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Poor spectral resolution inherent with cochlear implants diminishes the perception of frequency-place information. However, temporal resolution remains intact, and therefore research efforts have focused on temporal structure to improve the functionality of auditory implants. Three types of temporal structure have been described: envelope, periodicity, and fine structure. Investigating the perceptual importance of temporal structure in speech requires recovering these separate forms of structure for presentation. The common technique has been to use auditory chimera (AC) processing [Smith, Delgutte, &Oxenham, 2002, Nature, 416, pp. 87-90], which uses the Hilbert Transform amplitude envelope and instantaneous frequency signal with a filter bank. However, AC processing is only one of an unlimited number of ways to represent the temporal information in speech. Pitch synchronous (PS) signal processing for voiced portions of speech provides an alternative method to represent the gross temporal envelope (GTE), periodicity (or source signal), and fine structure or gross spectral envelope (GSE). The GTE can be represented by the relative amplitude across individual pitch periods in an utterance. The GSE can be represented by the relative amplitude across harmonics in a pitch period. The source signal can be represented by a signal with equal amplitude across each pitch period, and equal amplitude across harmonics.

In this work, three techniques were developed to recover and modify the temporal information of a speech signal. PS Temporal Amplitude Normalization was developed to recover and modify the GTE by scaling the amplitude of individual pitch periods separately. PS Spectral Amplitude Normalization was developed to recover and modify the GSE by scaling the amplitude of individual harmonics in each pitch period. A method of recovering the source signal was developed to remove the GTE and GSE. Using previously developed methods, the periodicity of the signal can be modified by resampling each pitch period, altering the fundamental frequency. The first of the three techniques above was used in a perceptual experiment to compare processing methods in which listeners were presented with three conditions of stimuli: unprocessed, PS processed and AC processed. Percent correct syllable recognition was calculated for each condition of stimuli. Also, listeners labeled stimuli for perceived naturalness on a five-point Likert scale. PS processed stimuli were rated as sounding as natural and labeled as accurately as unprocessed stimuli. AC stimuli were rated as sounding less natural and were labeled less accurately.

In conclusion, PS processing provides an alternative method of describing, recovering and modifying the temporal information in speech signals. These techniques have many applications for perceptual experiments.

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R20: A SOFT MASKING STRATEGY FOR SIMULTANEOUS SUPPRESSION OF NOISE AND REVERBERATION IN COCHLEAR IMPLANTS

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Although most cochlear implant (CI) users identify words and sentences with high accuracy in quiet and anechoic environments, their ability to recognize words degrades significantly in the presence of reverberation or/and noise. Reverberation and noise mask the speech in a rather complementary fashion. Hence, it is not surprising that the combined effects of reverberation and noise are more detrimental to speech intelligibility than that of either reverberation or noise-alone. The adverse impact of noise or reverberation on speech quality and intelligibility can be partially mitigated in the absence of the other. However, the suppression of both reverberation and noise in noisy reverberant environments is a challenging task, as the noise is added to the convolutive distortion caused by reverberation.

There exist practical algorithms for noise or reverberation suppression in CIs. Nevertheless, algorithms capable of tackling both reverberation and noise are more complicated and have not yet been investigated in the context of speech intelligibility improvements for CI users.

In the present study, we investigate a blind single-channel soft masking strategy to simultaneously suppress the negative effects of both reverberation and noise on speech identification performance of CI users. In this strategy, noise power spectrum is estimated from the non-speech segments of the utterance while reverberation power spectrum is computed as a delayed and scaled version of the reverberant speech spectrum. Based on the estimated noise and reverberation power spectra, a weight between 0 and 1 is assigned to each time-frequency (T-F) unit to form the final mask.

To evaluate the proposed speech enhancement strategy, intelligibility listening tests have been conducted with CI users for two reverberation times (T60 = 0.6 s and 0.8 s), wherein most of the existing dereverberation techniques fail to improve intelligibility, at a signal-to-noise ratio (SNR) of 15 dB (speech-shaped noise). A PDA research platform was employed to present the stimuli to CI subjects. In the implementation of the algorithm, the same parameters (frame size, frame shift, etc.) as the ones used in the ACE strategy are utilized. Obtained results indicate substantial improvements in speech intelligibility in both reverberant-alone and noisy reverberant conditions considered.

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R21: ENHANCING RAPID VARIATIONS OF COCHLEAR IMPLANT SIGNAL USING A SIMPLE MODEL OF SHORT-TERM ADAPTATION

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Much speech information is embedded in signal intensity variations within and across spectral channels. In cochlear implants slower temporal envelope variations provide the most usable cues to dynamic speech features, since variations of temporal fine structure are either not available or cannot be detected by the users. The present study evaluates the hypothesis that emphasizing within channel envelope variations improves implantees' access to dynamic speech features.

Previous attempts to emphasize envelope cues used onset enhancement techniques to improve perception of rapid intensity changes in a short temporal window and showed some improvement in the perception of speech in quiet and noise (Geurts and Wouters 1999; Vandali 2001). The method adopted in this study utilizes a linear high-pass filter resembling the short-term adaptation observed in the auditory nerve responses to acoustic stimulation. The motivation for this approach is that auditory nerve adaptation, at least partially, is attributed to reduction in hair cell transmission, which is absent in electrical stimulation. Thus applying an adaptation-like process to the envelope of the signal at speech processor output might produce a more natural neural response to dynamic speech features by emphasizing instances of electrical stimulation with rapid intensity changes (increments and decrements) relative to segments with more steady intensity. The adaptation model was characterized by two parameters: decay time constant and onset to steady-state ratio, and was applied uniformly to the signal of different implant electrodes after adjusting for the electrode's dynamic range.

Perception of consonants and sentences in quiet and background noise was evaluated and compared to that of the non-processed signal while several parameters of the adaptation model were applied. An improvement between 5 and 10 percent was observed in both tasks in most tested subjects with an adaptation model that had a time constant of 25ms or 50ms. The improvement reduced at high levels of background noise. Analysis of consonant features showed that transmission of temporal and spectral information improved by the new processing. Further improvement in consonant tests in quiet was observed with the processed stimuli after presenting to the subjects several training blocks where feedback was provided on the correct consonant. To compare effects of envelope enhancement on within and across channel features, perception of consonants with and without the new processing was evaluated while a single electrode was activated with the overall temporal envelope of the signal as in a single channel implant. A consistency was observed between single and multiple electrode conditions in the model parameters that provided the highest benefit. The results suggest that emphasizing envelope variations on implant electrodes by applying a model of short-term adaptation has the potential to improve implantees' access to within and across channel speech cues.

R22: RELATIVE STRENGTH OF PLACE PITCH CUES RESULTING FROM CURRENT STEERING AND FROM ELECTRODE TRANSITION EFFECTS

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Improved encoding of voice or musical pitch is a fundamental objective in cochlear implant (CI) research. The number of discriminable place pitches is limited by channel interaction. Studies considering current focusing and current steering to improve CI spectral resolution have demonstrated increases in the number of discriminable pitches.

However, a factor intrinsic to multichannel CI stimulation may confound efforts to create additional virtual channels through current steering. CI speech processors typically filter input signals into overlapping frequency bands, with each electrode in an array associated with a particular frequency band. An input pure tone presented at the centre frequency of a given band results in one electrode being stimulated maximally, while adjacent electrodes receive smaller stimuli. As the frequency of this tone increases, this same set of electrodes are activated for a range of frequencies with only their relative stimulus intensities changing. This has a current steering effect that leads to better than expected frequency discrimination (Pretorius &Hanekom, 2008, Hear. Res. 244, 77-84).

However, at specific frequencies, an additional electrode at one of the extremes of the set of active electrodes is activated, or one electrode is deactivated. We call this an electrode transition effect. These will continuously occur when a CI user listens to speech or music. This addition or removal of an electrode from an activated set should result in a pitch shift, and these pitch shifts may conceivably be larger than those achieved by current steering. This may negate the effectivity of virtual channels achieved by current steering. The present study investigated the relative importance of current-steered pitch changes and pitch shifts resulting from electrode transitions.

Nine adult CI users participated in experiments. Stimuli were designed to produce pitch changes through current steering or via electrode transitions. Eleven conditions were tested, repeated on an apical and a basal set of electrodes. Three conditions steered current by using graded stimulation levels on two adjacent electrodes. Six more conditions effected electrode transition stimuli by adding low level stimulation on a third electrode located basally or apically of the original two. Two reference conditions activated either just the most basal or most apical electrode in these sets. All stimuli were carefully loudness balanced. Pitch ranking to determine which stimuli resulted in the most prominent pitch changes followed a method of Long et al, 2005 (Ear. Hear. 26, 251-262). Results were compared to electrode discrimination data in the same CI users.

Results varied considerably between participants. Some perceived pitch changes in electrode transition conditions, but not in current steering conditions. The opposite was true for a second group. A third group was sensitive to both types of changes. In general, CI users with poor electrode discrimination abilities were insensitive to both types of pitch cues. Results suggest that current steering techniques will only be successful in some CI users, but not all, and perhaps only in certain parts of the electrode array. Applying current steering techniques generically on commercial processors may result in lower than expected success rates.

R23: PARTIAL TRIPOLAR STIMULATION IMPROVES SPEECH IN NOISE AND SPECTRAL RESOLUTION

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Although each electrode in a cochlear implant provides independent information, the spread of excitation from adjacent electrodes overlaps such that multiple electrodes stimulate the same neural populations. Current focusing techniques, where multiple electrodes provide simultaneous stimulation out of phase to restrict the spread of current, have been proposed to reduce channel interaction and spread of excitation. One current focusing technique, partial tripolar stimulation (PTP) involves stimulating on two flanking electrodes in the opposite phase of a central electrode. Previous work has shown that the spread of excitation from PTP stimulation can be narrower than monopolar (MP) stimulation. In the present study, we implemented MP and PTP stimulation in novel speech processing strategies. Patients were tested on speech in noise and spectral resolution tasks to evaluate differences between MP and PTP speech processing strategies.

Two novel sound coding strategies were implemented in an Advanced Bionics body worn speech processor. Each strategy consisted of 14-channel CIS on electrodes 2-15. The two strategies were identical in all parameters except for stimulation mode (MP stimulation for Exp-MP strategy and PTP for Exp-PTP strategy). M-levels for each electrode were carefully loudness balanced across the two strategies to ensure an equivalent spectral shape. Performance with each strategy was evaluated with HINT sentences, digits in noise and a new test to measure spectral resolution, the Spectral-temporally Modulated Ripple Test (SMRT). The SMRT consists of a spectral ripple whose phase drifts in time, eliminating potential local loudness or edge effects of traditional spectral ripple tasks.

The average improvement in Speech Reception Thresholds (SRTs) with Exp-TP was 2.7 dB for HINT sentences in noise and 3 dB for digits in noise. All subjects performed better with Exp-TP with HINT sentences. One subject performed similarly with both strategies for the digits in noise task while all others performed better with the Exp-TP strategy. All but one subject demonstrated better SMRT scores (i.e. better spectral resolution) with the Exp-TP strategy. Current focusing (specifically PTP stimulation) provides better speech understanding in noise as well as better spectral resolution. Although further optimization of the strategies are needed, current focusing is a promising tool for better performance in future cochlear implant strategies.

R24: PROGRESS AND CHALLENGES IN HUMAN VESTIBULAR IMPLANTATION

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Four human subjects with uncontrolled Meniere's disease have undergone implantation of the UW/Nucleus vestibular neurostimulator and longitudinal assessments of between six months and two years. These have included audiometry and vestibular testing, as well as electrically evoked eye-movements, postural measures, perceptual assessments and ECAPs. These results have been compared with those from chronically implanted normal rhesus monkeys. Notable differences between the human and animal subjects include the failure to preserve hearing and vestibular function in human, as well as marked fluctuation of electrical excitability as demonstrated by eye-movements, perception and ECAP measures. We hypothesize that these fluctuations, as well as the greater sensitivity to labyrinthine injury in the human are due to the presence of Meniere's disease. Humans also demonstrate an unexpected divergence of physiological and perceptual outcomes that have profound implications for treating vestibular disease with neurostimulation technology. Detailed study of this divergence will be needed to apply results of animal studies to development of a clinically useful vestibular prosthesis.

R25: IMPROVED PERCEPTION OF MUSIC AND MANDARIN TONES WITH A HARMONIC CODING STRATEGY FOR COCHLEAR IMPLANTS

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Harmonics are abundant in speech and music sounds and they carry important information for perceiving pitch and segregating sounds. However, due to the inherently coarse spectral and temporal resolution in electric hearing, the question of how to deliver harmonic information to cochlear implant users remains unresolved. In this study, a harmonic-single-sideband-encoder (HSSE) strategy is proposed that explicitly tracks the harmonics in speech and music and transforms them into pulse stimuli carrying pitch and temporal fine structure information. The fundamental question on how many harmonics are necessary for speech and music perception, however, has not been thoroughly investigated. The current study tentatively addresses this question in normal hearing listeners, and then evaluates the effectiveness of HSSE in cochlear implant users.

To investigate the contributions of harmonics to auditory perception, a coherent demodulation technique was used to manipulate the number of harmonics in speech and music signals. Listening experiments were conducted with normal hearing listeners using synthesized sounds containing a subset of harmonics. Their performance on phoneme recognition and melody identification was measured as a function of the number of harmonics tracked. Based on findings from the listening experiments, a limited number of harmonics were selected to be encoded in the HSSE strategy. The HSSE strategy was then evaluated with eight cochlear implant users. Their ability to identify melodies and recognize musical instruments was evaluated using the Clinical Assessment of Music Perception (CAMP) tests. In addition, Mandarin tone recognition with the HSSE strategy was assessed in one Mandarin-speaking cochlear implant subject.

Our experimental results indicate that 8-12 harmonics are needed to reach relatively normal speech recognition and musical timbre recognition. For cochlear implant subjects with 8-12 harmonics, a significant advantage of HSSE over subjects' clinical strategy was demonstrated with musical timbre recognition: the largest improvement was 35% and the mean 17% ($p<0.013$). On melody recognition, two cochlear implant subjects showed 20% improvement with HSSE; however, the mean improvement of 7% across subjects was not significant ($p>0.090$). The Mandarin-speaking subject achieved about 15% better tone recognition performance with the acutely-fitted HSSE strategy. English speech perception was neither enhanced nor degraded by HSSE.

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R26: ACES: A NEW SPEECH CODING STRATEGY FOR COCHLEAR IMPLANTS WHICH COMBINES THE BENEFITS OF CIS AND ACE CODING STRATEGIES

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Continuous interleaved sampling (CIS) and Advanced Combination Encoder (ACE) are the two most widely used speech coding strategies in the cochlear implant industry. In the CIS strategy a fixed number of envelopes are computed and only the corresponding electrode sites (typically 8-10) are used for stimulation. Whereas in the ACE strategy, only a subset N (typically N varies between 8 and 12) out of M (22 in the Nucleus device) envelopes are selected based on maximum amplitude and used for stimulation. In this way all M electrode sites are utilized for stimulation. The maxima selection criterion used in the ACE strategy works well in quiet and in relatively good SNR environments as it captures the perceptually relevant features of speech such as formant peaks. However, this criterion could be problematic in some instances for various reasons. First, at low SNR's the selected amplitudes could include information from the masker-dominated channels, thereby confusing the listeners as to which is the target and which is the masker. Second, the selection is done all the time for all segments of speech, including the low-energy segments where noise will most likely dominate and mask the target signal. Third, the maximum criterion may be influenced by the spectral distribution of the target and/or masker. If, for instance, the masker has high-frequency dominance, then the selection will be biased toward the high-frequency channels.

In contrast CIS is not influenced by the spectral distribution of the target and/or masker as all the frequency channels are stimulated in each time frame so no part of the signal is missing as in ACE due to N-of-M maxima selection. However part of the frequency resolution is lost due to stimulation on typically 8-10 fixed electrodes. Further, CIS introduces noise from the whole signal spectrum compared to the ACE where noise from only N number of channels is introduced.

Clearly, a better selection criterion or a modification to the current algorithms may compensate for the above shortcomings of the current coding strategies. We present a new coding strategy named ACES (ACE+CIS) which tries to combine the individual benefits of ACE and CIS, while avoiding some of their shortcomings. The new speech coding strategy provides the whole signal for stimulation instead of N out of M channels without compromising the better frequency to electrode/place mapping of the ACE coding strategy. The results of speech understanding performance in noise for CI subjects compared to the ACE strategy will be presented.

This work is supported by the EU ITN-project AUDIS and Cochlear Technology Centre Belgium.

R27: PITCH DISCRIMINATION OF AMPLITUDE-MODULATED CURRENT PULSE TRAINS WITH HARMONIC RELATED CARRIER PULSE RATE.

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Amplitude modulation of biphasic pulse trains is known to provide some degree of pitch perception in cochlear implant subjects. Especially in tonal languages, pitch perception is important for speaker identification, prosody and speech intelligibility. Experiments that were performed with sinusoidally amplitude-modulated (SAM) pulse trains at different carrier pulse rates indicated a benefit for pitch perception conveyed at carrier rates that were harmonic to amplitude modulation frequency (MF) over inharmonic carrier rates [McKay CM, McDermott HJ, Clark GM. J. Acoust. Soc. Am., 96, 2664-2673]. The aim of the present study was to investigate the effect of harmonic carrier rates on pitch perception of SAM pulse trains as a means to enhance pitch information in future speech coding strategies.

A two-interval, two alternative forced-choice task following weighted up/down rules was used to measure just noticeable pitch difference (JNPD) between a MF of 100 Hz and higher. The carrier rate was either fixed at a constant value of 955 pps or was set harmonic to an integer multiple of the MF (e.g. 100 Hz at 900 pps and 120 Hz at 1080 pps). To avoid possible loudness cues, the amplitude of the pulse trains was roved by 5%. The pulse trains were presented either on a single electrode (i.e. 19) or on four electrodes (i.e. 7, 11, 15 and 19). To compare these results with previous data, JNPDs of simple unmodulated pulse trains of 100 pps and above on a single electrode were assessed to compare amplitude modulation with pure temporal pitch.

Up to date, three cochlear implant patients were tested. Preliminary results indicate a decrease of JNPD for the one electrode condition and an increase of JNPD for the four electrodes condition when using harmonic carrier rates compared to unharmonic ones.

Further experiments will increase the number of tested subjects and add additional conditions with additional electrodes and different harmonic stimulation rates for individual electrodes in the multi-elelctrode SAM pulse train condition.

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R28: AN OBJECTIVE EVALUATION OF THE DE-REVERBERATION EFFECT OF COCHLEAR-IMPLANT AND HEARING AID FRONTEND ALGORITHMS

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The effect of reverberation on speech intelligibility (SI) in cochlear implant (CI) listeners was investigated recently [Hazarati and Loizou, 2012, Int. J. Audiol., 51(6), 437-443]. It was shown that (1) CI users' SI performance is considerably reduced even for moderate levels of reverberation and that (2) the detrimental effect of reverberation becomes even more severe when combined with moderate levels of background noise. In hearing aids a variety of frontend signal processing features is employed to suppress unwanted noises. An important research question is whether some of these frontend algorithms are potential candidates for counteracting the detrimental effect of reverberation on SI in CI users [Chung, Nelson and Teske, 2012, Hear Res, 291(1-2), 41-51].

We objectively tested the feasibility of sound processing algorithms for an application to reverberation cancellation. The algorithms included two frontend processing approaches: a beam-former (BF) and the EchoBlock reverberation cancellation method. The AB ClearVoice coding strategy was also tested. Reverberant conditions were generated using a computer simulation based on the source-image method, including head-related-transfer-functions (HRTFs) and behind-the-ear (BTE) microphone characteristics. Three different rooms were simulated with the frequency-averaged RT60 varying from 0.4 to 2.6 seconds. Different direct-to-reverberant ratios (DRRs) were investigated. By employing a shadow-filter model, direct-path and reverberant-speech components were processed separately with attenuation values generated by the algorithm at hand. A signal-based, least-squares normalization technique [Naylor, Gaubitch and Habets, 2010, J. Elec. Comp. Eng, 2010, 1-5] was used in order to obtain an estimate for the DRR before and after processing.

Across all algorithms under investigation, the BF provided the largest estimated DRR-improvement; up to 4 dB. This was largest for negative room DRR, i.e. simulations with the listener situated outside of the critical-distance. These situations may be particularly difficult for CI subjects. BF and EchoBlock combined, provided similar benefits to the BF alone. Based on these outcomes, we hypothesize that the frontend algorithms mentioned above are beneficial for CI subjects in reverberant listening conditions. An ongoing study has enrolled 10 adult CI users. The study will assess speech perception in the laboratory set-up, as well as subjective feedback following a take home period. Data will be presented and discussed in relation to the objective findings.

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R29: COCHLEAR IMPLANT NOISE REDUCTION: PERFORMANCE, QUALITY AND ACCEPTANCE

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Noise reduction (NR) technologies in cochlear implants (CIs) sound processors have provided significant improvements in speech perception, as well as listening quality. Due to differences in CI and hearing aid speech understanding performance and stimulation modality, it is expected that hearing aid NR strategies will not be optimized for CI users. Recent psychoacoustic studies have demonstrated significant differences, and suggested changes optimising NR techniques specifically for CIs. This studies objective was to investigate the speech understanding performance, listening quality, acceptance and fitting of a NR technology designed for CIs.

CI recipients were tested using a NR technology designed for CIs compared to a standard NR technology and the ACE alone program. The Everyday SmartSound (ADRO and ASC) program was used as the baseline signal processing for all three test conditions. Word understanding in quite and sentence understanding in noise were used to assess speech understanding performance. Noise annoyance rating and overall quality rating were used to test listening quality. Speech perception and listening quality were tested in speech weighted noise, 20-talker babble and 4-talker babble. Take home testing was also performed to assess acceptance in a range of listening environments.

Acute clinical testing of a CI designed NR technology found significant sentence understanding improvements in speech weighted noise compared to the ACE program by 19% ($p<0.001$) and the standard NR strategy by 7% ($p<0.05$). The CI designed NR strategy also provided significant improvements in babble noise when compared to the ACE program ($p<0.001$). No intelligibility degradation in quiet was found. Noise annoyance rating was significantly reduced compared to the ACE program by 54 percentage points ($p<0.001$) and a standard NR strategy by 24 percentage points ($p<0.01$) in speech weighted noise. Overall quality rating was significantly improved over the ACE program by 49% ($p<0.001$) and a standard NR strategy by 19% ($p<0.01$). Significant quality improvements were also found in babble conditions.

This study has shown significant improvements in speech understanding and listening quality with a NR technology designed for CIs. The fitting and acceptance of this new technology has also been very positive for both clinicians and recipients respectively.

R30: TWO NOVEL CODING STRATEGIES BASED ON NEURAL EXCITABILITY

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Present day coding strategies are usually based on signal transmission concepts originally developed for communication systems, often emphasizing the appropriate processing and conditioning of the input signal. However, the neurophysiological properties of the stimulated neurons, which affect the capacity of these neurons in conveying the encoded information, are seldom taken into account.

The input signal is typically divided into multiple frequency channels, and the stimuli to be presented are selected based on the energy content in these channels. Often, the selected channels are clustered together, resulting in the stimuli being concentrated around specific locations along the electrode array. The neural population stimulated by such successive stimuli close together in space and time will generally not be able to respond to each and every stimulus due to the refractory behavior or excitability of these neurons. A coding strategy which accounts for this neural excitability would be aimed at minimizing such redundancies, thereby improving the effectiveness of the presented stimuli.

Specifically, stimuli presented during the absolute refractory period of a given neural population are not effective and the corresponding time interval is better utilized if these stimuli are presented at other locations that are able to react more effectively. Using a combined model of the electric field spread as well as refractory properties to determine the excitability of a neural population at any given point in time, two new coding strategies were developed and tested.
The first strategy, called Refractory State Coding (RSC), is effectively able to reduce both spatial and temporal clustering. Initial laboratory tests involving recognition of aCa consonant logatomes with RSC with 9 CI listeners demonstrated that RSC was not significantly poorer than the listeners' standard ACE strategy. At this point of the assessment, the RSC output had not yet been optimized, for instance, for loudness compensation due to the relative sparseness of the RSC activity compared to ACE which led to a reduction in perceived overall loudness.
The second strategy, called Excitability Controlled Coding (ECC), encodes the signal intensity on a given channel into the stimulation rate instead of the stimulation level. ECC therefore encodes loudness by varying the stimulation rate, while keeping the stimulation level constant. One aim of ECC is to minimize spatially-related channel interaction arising from the electric field spread, since the stimulation level, which is assumed to be the primary parameter related to electric field spread, will not increase with increasing signal intensity.

Both RSC and ECC have been successfully implemented within a Matlab application and will be presented here. The outputs and the results of initial assessments will be discussed.

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R31: IMPROVING COCHLEAR IMPLANT OUTCOME THROUGH ALTERED AUTOMATIC GAIN CONTROL AND COMPRESSION SETTINGS

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To deal with the limited dynamic range of electric hearing, most current cochlear implant (CI) systems use a single broad-band AGC system at the front-end of the sound processor. In contrast, acoustic hearing aids typically use multiple independent AGC channels. In experiment 1, we examined whether two-channel AGC would bring benefit to CI recipients.

A low-frequency noise was added to the 20 tokens in a vCv test, using an adverse signal-to-noise ratio. Four Advanced Bionics recipients listened to the stimuli, following processing by one- or two-channel AGC systems. Five normal hearing (NH) adults were also tested, using six-channel vocoded stimuli. All participants scored better with the two- than with the one-channel AGC. Group mean scores for two- and one-channel compression, respectively, were: 61% and 29% ($t=2.13$, $p=0.002$) for the NH group and 54% and 26% ($t=3.18$, $p=0.001$), for the CI group. Similar results were obtained with several real-world noises.

CI systems additionally use a near-instantaneous compressive “mapping function” to transform the envelope magnitude in individual channels to electrical stimulation current. In experiment 2, we examined the effect of different balances of front-end single-channel AGC and back-end compression. The Speech Reception Threshold (SRT: the signal-to noise ratio required for 50% intelligibility) was estimated for target sentences whose level varied between 50 and 80 dB SPL. SRTs were compared for three different fittings: (a) a clinical fitting where threshold (T) levels were set to 10% of most comfortable (M) levels, (b) a fitting with behavioral T levels and (c) a fitting with behavioral T levels with a lowered front-end AGC threshold and reduced near-instantaneous compression. Preliminary results indicate similar SRTs for conditions (a) and (b), but a lower (better) SRT for condition (c), by about 3 dB.

In summary, the results indicate that two-channel front-end AGC can give better results than one-channel AGC, and that, for a one-channel AGC system, performance can be improved by reducing the compression threshold of the front-end AGC and decreasing the near-instantaneous compression.

R32: FINE STRUCTURE INFORMATION BENEFIT FOR SOUND QUALITY AND SPEECH IN NOISE INTELLIGIBILITY IN BIMODAL COCHLEAR IMPLANT USERS

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Commonly used envelope-based cochlear implant (CI) speech processing strategies (e.g. CIS) do not provide access to low-frequency waveform fine structure or to interaural time difference (ITD) based spatial unmasking. A fine structure processing strategy (FSP), which attempts to transmit low-frequency waveform information in addition to high-frequency envelope information, might provide bimodal CI users with enhanced ITD cues and improve speech-in-noise perception, and sound quality. The present study investigated the effect of the fine structure information provided by the current MedEl FSP strategy (FS4) versus the alternative CIS strategy (HDCIS) on sound quality and hearing in noise in 11 adult bimodal CI users, with bilateral severe to profound hearing loss, at six months post CI activation. All participants used OPUS 2 (MedEl) processors, which were programmed with both strategies. All the tests were performed in both bimodal and CI alone configurations, and with both FS4 and HDCIS strategies.

For sound quality, the participants were asked to use a visual analog scale to rate short recordings as the listening configuration and CI strategy were varied. This was repeated for six recordings (rock music, classical music, and four different speech passages). Stimuli were presented in a sound booth using specially designed headphones and a computer controlled implant remote control that allowed trial-to-trial blind selection of the CI program. The results showed a significant overall sound quality preference for FS4 versus HDCIS and for the bimodal versus CI alone configuration, and also a significant interaction between the recordings, the configuration and the strategy used. Across all four speech recordings, the hearing aid provided no additional benefit with FS4, and it significantly improved quality when HDCIS was tested. This bimodal advantage was the opposite with rock music: it was present with FS4 and not significant with HDCIS. There was no significant quality difference between strategies when listening to classical music, although the bimodal configuration was preferred.

For spatial unmasking, 20-sentence HINT speech reception thresholds were obtained in an anechoic chamber with speech and noise co-located in front or with 900 spatial separation (speech in front and noise on the hearing aid side). The hearing aid was in an omnidirectional setting. A significant interaction between the CI strategy and the listening configuration was demonstrated; spatial unmasking was significantly higher for the configurations that provided low-frequency fine structure (FS4 alone, bimodal FS4, and bimodal HDCIS), than for HDCIS alone. This is consistent with the sound quality scores obtained for speech.

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R33: COMBINING HARMONIC REGENERATION WITH NOISE SUPPRES-SION TO IMPROVE SPEECH RECOGNITION BY MANDARIN COCH-LEAR IMPLANT LISTENERS

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Listening in background noise still presents difficulties to cochlear implant (CI) users. Although as a pre-processing based approach, noise suppression has been shown to be efficient and effective in improving CI listeners' speech perception in noise, there is an often-ignored issue: most noise reduction methods use preset optimization criteria and offer no remedy for the problem that the noise-suppressed speech no longer possesses the properties mandated by the acoustic process of speech production (for instance, the harmonic structures in voiced segments of the noise-suppressed speech are usually severely distorted).

There exist a small number of studies on the contribution of noise reduction to speech recognition by Mandarin CI listeners. These studies showed benefits of noise reduction similar to those for English-speaking CI users, i.e., noise reduction improves speech perception by Mandarin CI listeners. However, due to the fact that Mandarin being a tonal language relies on lexical tones to deliver sentence intelligibility, we hypothesize that noise-reduction algorithms combined with harmonic regeneration will provide further benefits by improving the representation of lexical tones, in particular for those Mandarin CI users who have access to low-frequency sound via hearing aids; in addition, regenerating harmonics may improve the formant representation of Mandarin vowels as well.

In the present study involving both testing Mandarin CI subjects that wear only CIs and testing normal-hearing listeners using acoustic simulation imitating electroacoustic stimulation, we propose a noise-reduction algorithm using the log-minimum mean square error (LogMMSE) criteria combined with a statistical-approach-based harmonic-regeneration algorithm that attempts to estimate the fundamental frequencies and magnitudes of the harmonics. Testing results for the tasks of Mandarin tone and vowel identification, and speech recognition showed significant improvements compared to those obtained with unprocessed noisy speech and speech processed by the LogMMSE algorithm alone.

R34: IMAGE-GUIDANCE FOR CUSTOMIZING COCHLEAR IMPLANT STIMULATION STRATEGIES

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Cochlear implants (CIs) are arguably the most successful neural prosthesis to date. Despite this success, a significant number of CI recipients experience marginal hearing restoration, and, even among the best performers, restoration to normal fidelity is rare. We have developed CT image processing techniques that can be used to detect the positions of implanted CI electrodes and quantify the electrode-to-neural interface for individual CI users. These techniques permit development of new, patient-customized CI processor programming strategies. In this poster, we will present one such strategy that uses this patient-specific spatial information to decrease cross-electrode channel interactions, and we will show that this new strategy improves hearing outcomes in a study conducted with 17 CI recipients.

Prior to this study, each of the 17 experienced CI users had reached asymptotic hearing performance via traditional behavioral programming approaches. Hearing performance was quantified for CNC words and phonemes, AzBio sentences in quiet and noise (+10 dB SNR) and BKB-SIN sentence recognition in noise. In addition to the standard measures of speech recognition, we assessed spectral modulation detection (SMD) at 0.5 and 1.0 cycle/oct using a method of constant stimuli. For each subject, hearing performance was assessed with his/her existing clinical program prior to remapping. Subjects used the experimental map based on our individualized position-based model for a period of 3-4 weeks after which hearing performance was again assessed. The experimental maps were implemented unilaterally for the 17 bilateral participants. Subjects demonstrated considerable improvement on all measures with average performance increasing from 22% to 33% (CNC words), 44% to 51% (CNC phonemes), 31% to 43% (AzBio), 21% to 34% (AzBio + 10 dB), 20 dB to 17 dB SNR (BKB-SIN), 40% to 49% (SMD at 0.5 cycles/octave), 25% to 44% (SMD at 1.0 cycle/octave), and subjective benefit of 11-percentage points (APHAB background noise scale). Performance in the non-treated ear for the 17 bilateral participants was unchanged across the two test points.

We believe these results are significant because: (a) this new strategy is compatible with the majority of existing cochlear implant systems; (b) it drastically improves hearing outcomes independent of device type; and (c) we believe this opens the door to a new avenue of research into other more comprehensive image-guided programming strategies that could further improve hearing outcomes with cochlear implants.

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R35: BEHAVIOURAL EVIDENCE OF NOISE-ENHANCED INFORMATION TRANSMISSION

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Whilst there is behavioural evidence that endogenous sensory noise can improve detection for hearing there has been no behavioural evidence that sensory noise improves the ability to discriminate between speech stimuli. A behavioural demonstration of noise-enhanced information transmission has been hindered by an inability to manipulate sensory noise in vivo. Moreover, since normal sensory systems are already intrinsically noisy, it has not been possible to show that information is optimized by a non-zero noise level. Here we use cochlear implant stimulation with human participants who have a hearing loss as a tool to investigate neural coding.

The study involved 12 participants with the HiRes90K cochlear implant (Advanced Bionics Corporation) and we used the Advanced Bionics Streaming Interface (HRStream) to access the electrodes. We restricted the information-bearing signal to a single electrode such that stimulus identification was based solely on the temporal pattern of evoked spikes. The stimuli were five vowels that differed in the frequency of the lowest two formants, F1 and F2; pairs of vowels shared formants such that in most cases identification of one formant frequency would not uniquely specify an individual vowel. We removed all intensity and duration cues for vowel identification other than the formant frequencies. After slight training, cochlear implant users identified the vowels presented either without noise or with independent noise added to several electrodes. The electrodes would be expected to produce slightly different random noise at different locations along the cochlea, thus making stochastic resonance possible for suprathreshold signals.

From the confusion matrices we show that the mutual information is optimized by a non-zero level of noise and that this is associated with the optimal transmission of F1 information. We therefore demonstrate that the human hearing system has the requisite decoding mechanisms to exploit the stochastic encoding of sensory information at the periphery.

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R36: MONAURAL SPEECH ENHANCEMENT FOR COCHLEAR IMPLANT BASED ON PITCH ESTIMATION COMBINED WITH CASA BINARY MASK ESTIMATION

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Monaural speech enhancement plays an important role in increasing cochlear implant (CI) recipient performance, especially with respect to improved speech perception in noisy background. Monaural speech enhancement is also a difficult problem since there is no spatial information available. In this study, we propose a novel speech enhancement algorithm based on pitch estimation which is combined with CASA binary mask estimation. The pitch estimation is performed in the frequency domain, and the target harmonic structure and noise spectrum are also estimated in frequency domain. Next, we convert the re-synthesizing stage into a CASA (computational auditory scene analysis) problem, which is inspired by the fact that a CI users' sound perception is more consistent with the temporal envelope rather than frequency spectrum.

The detailed speech processing algorithm consists of three steps: First, we propose a method for pitch estimation based on long-short duration analysis of the associated spectrum, which obtains a higher harmonic resolution while maintaining the short-stationary feature of the speech signal. Second, the target speech spectrum and the noise spectrum is estimated. The target speech harmonic structure is estimated using a long frame spectrum along with the estimated pitch, and then the target speech spectrum is generated by convolving the harmonic peaks with the Gaussian shapes. The noise spectrum is calculated by convolving the remaining spectral peaks after extracting the harmonic peaks using Gaussian shapes. Third, once we have the estimated the speech and noise spectra, we convert the remaining part of the problem into CASA binary mask estimation. We choose the gammatone filter as the auditory filter, and rather than decompose the original time domain signal into a series of channels, we calculate the energy in each channel by multiplying each gammatone band with the previously estimated speech spectrum and noise spectrum respectively. The CASA binary mask is estimated by comparing the energy in each Time-frequency domain. Finally, the target speech signal is re-synthesized using the estimated CASA mask and the noisy speech.

Listening experiments show speech intelligibility improvement is obtained when the CI users are presented with simulated noisy speech followed by processing with our algorithm (speech shaped noise and babble noise, 0dB and 5dB).

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R37: ASSESSMENT OF SPEECH PERCEPTION IN EVERYDAY LISTENING SITUATIONS IN COCHLEAR IMPLANT USERS WITH SOUND REPRODUCTION VIA WAVE FIELD SYNTHESIS

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The assessment of speech perception in noise in hearing aid (HA) or cochlear implant (CI) recipients in clinical audiological diagnostics is typically carried out in setups containing only a few loudspeakers. In order to determine the impact of advanced signal processing algorithms in HAs and CIs on noise suppression, dynamic multi-band compression or beam forming, a more sophisticated measurement setup is demanded.

A multi-channel sound reproduction system containing 128 loudspeakers was developed and realized in the anechoic chamber of our department. The sound reproduction method 'Wave Field Synthesis' (WFS) is used to create realistic auditory environments. Speech reception thresholds (SRT) were measured in 12 normal hearing participants and 14 CI users (7 CI wearing a HA in contralateral ear, 7 bilateral CI, all CI processors CP810, Cochlear). Speech processor settings were either ˜everyday', static beamforming (ZOOM), or adaptive beamforming (BEAM). SRTs were determined with the Oldenburg Sentence Test (OLSA) in either (1) moving noise source condition or (2) multi-source noise field (MSNF) condition (4 noise sources). Speech was always presented from 0°. Noise was presented in either two conditions: quasi-continuous (Oldenburg Noise, OlNoise) or modulated (Fastl-Noise). All noise sources are generated with WFS.

Average SRT in the moving noise condition in the control group was -10 dB SNR (OlNoise) and -18 dB SNR (Fastl-Noise). Average SRT in both CI groups was -3 dB (OlNoise) and +2 dB (Fastl-Noise). With the ZOOM setting, a decrease in SRT of 4 dB and with the BEAM setting an even higher decrease in SRT of 7.5 dB compared to the everyday setting was observed on average in both CI groups. Average SRT in the MSNF condition in the control group was significantly lower than in both CI groups (difference 6 dB in OlNoise and 12 dB in Fastl-Noise). In Fastl-Noise, the bimodal group showed lower average SRTs compared to the bilateral group. In both ZOOM and BEAM setting, SRT decreased about 3 dB compared to the "everyday" setting. No additional benefit was observed with the BEAM setting.

The implementation of static beam forming algorithms in cochlear implant speech processors gives significant improvement in multi-source noise conditions. The implementation of adaptive beamforming strategies provides additional benefit in listening situations with a single moving noise source in the rear. WFS can serve as a sophisticated tool to assess potential benefits of advanced signal processing strategies.

R38: AN APPROACH TO DRUG DISCOVERY: SCREENING FOR SURVIVAL AND NEURITE PROMOTING ACTIVITY ON SPIRAL GANGLION NEURONS

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One key goal in hearing medicine is to develop interventions that will maintain survival and promote neurite regeneration from spiral ganglion neurons. Given the importance of spiral ganglion neurons for normal hearing and for the function ofcochlear implants, one would assume that thousands of drugs and chemicals had already been evaluated for survival and neurite promoting activity in the cochlea. This has not happened, however, for a logical reason: Evaluating just one drug or chemical in the ear at multiple concentrations and time points, for effects on hearing and cochlear anatomy is so labor intensive that it is simply not possible to test thousands of drugs or chemicals in hearing loss animal models. Unable to systematically push forward with unique or unknown compounds in the ear, the field has thus far mainly been occupied with factors that were first vetted in other regions of the nervous system. To focus in on interventions aimed at cochlear neurons, a new approach is required to speed up the process and strike a productive balance between discovery and in vivo evaluation.
To this end, we have developed a novel prescreening procedure to evaluate thousands of compounds in a semi-unbiased manner. The goal is to narrow down a huge field of potential chemicals to a handful of the most promising to promote to in vivo evaluation.

Our approach uses a well-characterized and quantified primary culture system of dissociated spiral ganglion from newborn mice. From 8 mouse pups, 192 cultures are generated in 384 well plates, tested, immunolabeled, automatically imaged and quantified. Forty four chemicals and 16 controls can be assayed in quadruplicate using the material generated from 8 mouse pups.

In initial studies we demonstrated that the Rho kinase inhibitor H-1152 stimulates neurite growth in spiral ganglion cultures. We further demonstrated that other Rho kinase inhibitors, fasudil and hydroxyfasudil also stimulated neurite growth, albeit at a higher concentration. The assay is consistent and reproducible with both positive and negative controls. We have now screened the NCC clinical collection, a library of over 400 known bioactive compounds and have identified a class of compounds that stimulate the growth of spiral ganglion neurites in our in vitro assay. We have advanced several of these compounds for evaluation in a guinea pig model of noise induced hearing loss.

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R39: INSERTION, TRAJECTORY, ELECTRODE POSITION, HISTOLOGY, AND INTRACOCHLEAR TRAUMA RELATED TO THE ROUND WINDOW, PERI-ROUND WINDOW, AND COCHLEOSTOMY INSERTION OF STRAIGHT ELECTRODES

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Aim: To examine Cochlear America's new slim electrode and evaluate its performance in various approaches using real time fluoroscopy and post insertion histological sectioning.

Materials and Methods: Insertions using a new slim electrode were performed in fifteen fresh-frozen temporal bones. Five temporal bones were used for each of the three preparation methods: 1. cochleostomy (CO), 2. peri-round window (PRW), and 3. round window (RW). All insertions were recorded under fluoroscopy to analyze insertion angle and trajectory. Implanted arrays were embedded in resin, sectioned, and stained, to observe final electrode position and evaluate for insertion trauma.

Results: Successful insertion was achieved with all 15 electrodes. Cochleostomy insertions achieved a greater angle of insertion than PRW and RW insertions ($p < 0.05$). 14 of 15 insertions rested in the scala tympani and 1 RW insertion rested into the scalavestibuli.

Conclusions: The slim electrode design is suitable for all 3 approaches tested, but results were most favorable using a cochleostomy approach.

R40: AN ORGANIC MEMS FABRICATED BASILAR MEMBRANE SENSOR

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Work is underway in several labs on implantable artificial cochleae, where micro fabricated sensor structures are designed to mimic the tonotopic mapping function of the basilar membrane. In this paper, we report on a new design approach to develop a basilar membrane-like cochlear device to pick up the intracochlear pressure waves.

The travelling wave pressure adjacent to the basilar membrane peaks at frequency dependent locations, just as the basilar membrane motion does: the pressure carries tonotopic information. Our in-vivo pressure measurements show an elevated pressure gradient close to the basilar membrane, which is also predicted in cochlear computer models. With the aim of exploiting these frequency and spatially dependent characteristics of intra-cochlear pressure, we designed a piezoelectric polymer sensor based on the integrated organic MEMS fabrication technology. A 50µm thick PVDF film is used as both the piezoelectric material and the device substrate. Thin film Au layers are deposited and patterned to form the sensor electrodes. The test device is encapsulated with Parylene coating. As the first step of our study, an implantable prototype with two sensing sites was constructed and tested in-vitro. Each of the sensors is designed to have the same broad frequency range, to pick up the locally tuned pressure. To further improve the sensor performance, we will fabricate organic transistors on the same PVDF substrate to integrate the pre-amplifier with the pressure sensors and thus enhance the signal-to-noise ratio. This paper will present an evaluation of the prototype performance and the feasibility of this design concept.

R41: SOUND - TRANSMISSION EFFICIENCY IN ROUND WINDOW STIMULATION

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Introduction: Mechanical stimulation of the round window (RW) of the cochlea is successfully done with implantable hearing aids, but clinical outcomes are variable. The Direct Acoustic Cochlea Stimulator (DACS, Phonak Acoustic Implants) is intended to stimulate the cochlea directly by a piston, but can be used for RW stimulation also. Here the impact of static force applied to the RW on performance and output efficiency of the DACS is investigated.
Methods: In fresh human temporal bones (TBs) the maximum equivalent sound pressure output of round window stimulation was determined experimentally in analogy to the ASTM standard F2504.24930-1. Stapes footplate (SFP) displacement in response to sound applied to the external ear canal was determined using a Laser Doppler Velocimeter. Only ASTM compliant TBs were implanted with the actuator and contributed data to the analysis. The stimulation was performed perpendicular to the RW with a modified DACS actuator by a spherical tip (diameter 0.5mm) as interface to the RW. Starting at a position in front of the RW, the position of the actuator was moved with a micro-manipulator in increments of 50 micron into the RW membrane and the resulting axial static force was determined by a force sensor. At each position the actuator displacement output and the SFP in response to actuator stimulation was determined. From the SFP response to sound at the tympanic membrane and actuator stimulation the achieved equivalent sound pressure level was calculated. For averaging series of displacement responses, results were aligned using an axial static force threshold of 4 mN.
Results: Ten TBs within the acceptance range of the ASTM standard contributed data to further analysis and were very similar in stress-strain characteristics. At an axial static force of 3.90 +- 0.70 mN (MV, SD) an average sound equivalent sound pressure level of ~103-120 eq. dB SPL (@1Vrms) was found at low frequencies (< 4 kHz). At higher frequencies (6 - 10 kHz) the achieved output dropped to ~90 eq. dB SPL. This force was below the actuator safe operating limit (< 50 mN) and at least a factor 10 below the rupture limit of the RW seen in our experiments. At higher forces > 4 mN the transfer function from the RW to the SFP increased monotonically with the applied forces (> 40 mN), allowing even higher output by increasing the pretension.

Conclusion: The sound transmission efficiency to the RW crucially depends on the static pretension applied to the RW. That sound transfer efficiency monotonically depends on the static force is of great practical importance, not only for the DACS, but also for all other devices used for RW stimulation. At a moderate pretension the DACS actuator provided sufficient output for moderate and pronounced sensor neural hearing losses.

R42: DETECTING AND REMOVING REVERBERATION FROM COCHLEAR IMPLANT PULSE TRAINS

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Speech recognition degrades more rapidly for cochlear implant listeners than for normal hearing listeners in reverberant environments because cochlear implants provide more limited spectral and temporal information [Kokkinakis and Loizou, 2011, J. Acoust. Soc. Am., 130, 1099-1102]. The goal of this study was to improve speech recognition in reverberant environments by mitigating the effects of reverberation by directly manipulating the cochlear implant pulse trains. First, a reverberation detection algorithm was developed to control the processing when switching between the mitigation algorithm and the case where no mitigation was needed. Second, the potential benefit of mitigating two different effects from reverberation was assessed. Finally, a mitigation algorithm was developed and assessed.

There has been little investigation of reverberation detection in the literature; however, it would provide a meaningful method to control the application of a mitigation algorithm. Ideally, a mitigation algorithm would completely remove reverberation and would not affect non-reverberant signals; however, realistically most noise and reverberation mitigation algorithms make errors and it would be preferable to not apply them unless needed. Here, reverberation detection was carried out by developing features based on the frequency-time matrices generated by the ACE speech processing algorithm. A maximum a posteriori (MAP) classifier was able to successfully discriminate between speech in reverberation, in quiet, and in noise 94% of the time.

Reverberation effects can be categorized into two types: overlap-masking (the masking of one phoneme by a preceding phoneme) and self-masking (masking within an individual phoneme). The potential benefit of mitigating these two effects was studied by comparing speech recognition scores for speech in unmitigated reverberation to speech after ideal self-masking mitigation and after ideal overlap-mitigation. Testing was conducted with acoustic models in normal hearing listeners and with cochlear implant users listening through their devices. Results were analyzed to determine the severity of each effect.

Finally, a speech processing algorithm that mitigates overlap-masking effects was implemented that was initiated once reverberation was detected in a pulse train. Performance of the reverberation-mitigation algorithm was evaluated in simulation, comparing the pulses that the algorithm determined to be overlap masking with the known truth.

R43: COCHLEAR IMPLANT SPEECH PROCESSING STRATEGY BASED ON A MODEL OF THE HUMAN PERIPHERAL AUDITORY SYSTEM

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To date, there are quite a few cochlear implant (CI) speech processing strategies on the market each having its advantages and disadvantages. However, extensive training can typically make one strategy just as successful as another. At the same time, rehabilitation outcomes indicate that current systems have reached their peak potential: Many CI users can communicate without lip-reading, but speech perception may suffer greatly from adverse acoustical conditions. This confirms the need for better algorithms.

A different approach -with more potential, but also with more pitfalls- is to rely completely on auditory models. This way, important properties of the auditory system and psychoacoustic phenomena shall be included inherently. To make sure that we are not bound to old methods, we have created a novel sound processing strategy, SAM (Stimulation based on Auditory Modeling), from scratch. SAM incorporates active cochlear filtering (basilar membrane and outer hair cells) along with the mechanoelectrical transduction of the inner hair cells. SAM also provides the stimuli with stochastic properties to reduce periodic characteristic while keeping temporal fine structure.

The idea of using auditory models in CI signal processing is not new, but still largely unexplored. Unlike in preliminary studies (see e. g. Wilson et al., 2010), our strategy employs a model of cochlear hydromechanics and has been tested with implantees of the Cochlear(TM) system. Computational evaluation shows that aspects of normal cochlear processing that are missing in common CI strategies can be replicated by SAM. This is meant to improve overall CI user performance, which we have at least partly proven in a pilot study with five CI users.

This presentation elaborates on SAM's signal processing and summarizes outcomes of the computational evaluation and that of the pilot study with CI users.

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Reference: B. S. Wilson, E. A. Lopez-Poveda, and R. Schatzer, "Use of auditory models in developing coding strategies for cochlear implants," in Springer Handbook of Auditory Research, vol. 35, Computational Models of the Auditory System, R. Meddis, R. R. Fay, E. A. Lopez-Poveda, and A. N. Popper, Eds, Boston, Massachusetts, USA: Springer Science+Business Media LLC, pp. 237-260, 2010.

R44: CONSIDERATION OF TEMPORAL MASKING IN COCHLEAR IMPLANT SPEECH PROCESSING STRATEGIES: THE TPACE STRATEGY

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MP3000 is the first clinically applicable signal processing strategy for cochlear implants making use of a psychoacoustic masking model. The rationale for using masking mechanisms in signal coding for cochlear implants was to abandon perceptually irrelevant signal components from the audio signal and, by doing so, save bandwidth for the very limited electrode-nerve interface. However, so far only simultaneous masking effects had been considered, as this effect, from experience in normal hearing subjects, held the largest potential for saving bandwidth. The implementation allowed for a reduction of the number of electric stimuli without compromising hearing performance [1].

Recently, it was decided to pick up the idea of masking again and the original PACE (Psychoacoustic Advanced Combination Encoder) algorithm, the research version of MP3000, was upgraded to additionally account for temporal masking effects (TPACE). Two implementations with different time constants have been designed within the Nucleus Implant Communicator research environment for acute testing of subjects. For determining the exponential decay of the temporal masking effect, the value T was defined, which denotes the time when the masking effect is only half as effective compared to its strength at the time t0. The two Tvalues used here were 0.5 ms and 1.1 ms. Twelve subjects were recruited and tested in one single session with four lists of HSM sentences in noise in each condition. These results were compared to the baseline-condition PACE (basically T= 0), which was also assessed with 4 lists of HSM sentences in noise for each of the subjects. A statistically significant increase in performance was found for the condition with T= 0.5. Results were 35 % for PACE, 45% for TPACE with T= 0.5 and 39% with TPACE with T= 1.1 ms.

It seems that the consideration of short-acting temporal masking can improve speech understanding in noise in cochlear implant subjects. So far, only time-constants smaller than 10 ms have been investigated, while temporal masking in normal hearing subjects lasts up to 200 ms. The study is still ongoing and latest results and implications will be presented at the conference.

[1] Buechner A, et al.. Clinical evaluation of cochlear implant sound coding taking into account conjectural masking functions, MP3000©. Cochlear Implants Int. 2011 November; 12(4): 194-204.

R45: SPEECH INTELLIGIBILITY IMPROVEMENTS PROVIDED BY IDEAL TIME FREQUENCY MASKING NOISE REDUCTION ALGORITHMS IN COCHLEAR IMPLANTS

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In a previous study, we examined the potential for improving the intelligibility of noisy speech when using ideal time-frequency masks to perform noise suppression. The term"ideal" was used to indicate the perfect estimation of relevant parameters for the mask computation. We compared time-frequency masks based on binary gains (so-called ideal binary mask (IBM)) against continuous (soft) masks derived from the Wiener filtering approach (ideal Wiener filter (IWF)). With the noise reduction performed on the Bark-spectrum scale, the IWF vastly outperformed the IBM, both in terms of intelligibility improvement and resulting speech quality. Moreover, we demonstrated that the IWF was also much less sensitive to errors in parameter estimation, as compared to the IBM. We suggested the use of soft masks for noise suppression in auditory prostheses.

In this study, we compared the potential of the two approaches for speech intelligibility and quality enhancement for cochlear implant (CI) users. The investigation was conducted on normal-hearing (NH) listeners listening to noise vocoder CI simulations, as well as on CI users.

The potential for speech intelligibility improvement was assessed in a sentence recognition task with ideal mask estimates in multi-talker babble noise and with an interfering talker. The robustness of the approaches was evaluated by a sentence recognition task with simulated errors in the estimation of the mask parameters. Quality was evaluated in a two-stage, pair-wise preference-rating test conducted on the group of CI users.

While the IWF approach outperformed the IBM approach in NH listeners listening to noise vocoded sounds, there was no significant difference in scores obtained with CI users. For the latter group, both mask patterns yielded scores close to that obtained in a quiet condition. Furthermore, the estimation errors did not significantly degrade the speech intelligibility in CI users. In terms of quality, the IWF outperformed, slightly, the IBM processed signals.

The results suggest that the choice of the derivation of the mask pattern is not that crucial for current cochlear implants. There seems to be a reduced sensitivity to speech distortions among CI users. The study demonstrates that testing new strategies with CI users is indispensible because results obtained with NH listeners did not translate to similar results for CI users.

R46: ADAPTIVE LOUDNESS GROWTH FUNCTION

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The Nucleus 6 processor has three processing blocks that influence the dependence of stimulation current on sound level. Firstly, a tri-loop Automatic Gain Control (AGC) observes the RMS signal level, and controls the front-end gain. Secondly, ADRO applies slowly varying gain to the filter bank output envelopes. Thirdly, the Loudness Growth Function (LGF) defines the mapping from envelope levels to currents. Envelopes at the LGF saturation level are mapped to C-level current, and envelopes at the LGF base level are mapped to T-level current. Gain control interacts with the LGF: a 10 dB decrease in gain is equivalent to a 10 dB increase in both the LGF saturation level and the LGF base level. Thus AGC and ADRO shift the dynamic range up and down to suit the sound environment.

Our previous work demonstrated the feasibility of replacing front-end AGC with a multichannel AGC on the filter bank outputs. This brings together all the gain control elements at a single point in the signal path, and enables further simplification and optimization. In the present study, tri-loop AGC, ADRO and LGF were replaced by a new algorithm called Adaptive LGF (ALGF). ALGF controls two parameters. The peak signal level controls the LGF saturation level, and a noise floor estimate controls the LGF base level. Thus the dynamic range is both shifted up and down, and also expanded or contracted to suit the sound environment.

Six adult CI recipients participated in this study. Two processing conditions were compared: the Nucleus 6 signal path and ALGF. Both conditions were implemented on a Simulink-xPC system. Neither condition included noise reduction (SNR-NR or Wind Noise Reduction). Speech intelligibility was measured with an interleaved roving Speech Reception Threshold (SRT) test. Sentences alternated between three speech presentation levels, 50, 65 and 80 dB SPL, in a fixed sequence. The level of speech-weighted noise was adapted independently for each speech level, giving an SRT for each speech level. At 80 dB, all six subjects obtained better (lower) SRTs with ALGF (Wilcoxon signed-rank test $p = 0.03$). At 50 and 65 dB, there was no significant difference between processing conditions. These results demonstrate the feasibility of the new algorithm.

R47: DESIGN AND EFFECTS OF POST-SPECTRAL OUTPUT COMPRESSION IN COCHLEAR IMPLANT CODING STRATEGY

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Cochlear implant coding strategies classically integrate automatic gain control on the input audio stream, and then a front-end or output compression function is applied to optimize the acoustical to electrical dynamic range. This classic signal processing method aims to narrow the acoustical dynamic range to let more loudness quantization steps.

The goal of the present study is to describe the new signal processing strategy for the Neurelec's forthcoming speech processor, which integrates post-spectral analysis compression. A novel frequency selective compression has been used. Compression settings used a new paradigm that eases and speedups the audiologist tuning.

Three presets were designed to maximize the speech information sent to the patient while assuring comfort on noisy and loud conditions. These pre-sets were statistically determined in order to preserve 95% of the speech information in quiet, medium and loud environments.
10 patients were fitted with this output compression function and tested for loudness growth, pure tone thresholds and speech tests in quiet and in noise. Results showed improved intelligibility in quiet for loudest presentation levels and in noise, with observed loudness growth in normal range.

All together, these results supports the fact that output compression designed without automatic gain control allows a full control in dB from the fitting software and showed promising results.

R48: NOISE-REDUCTION ALGORITHM IN COCHLEAR IMPLANTS: DESIGN, AND OUTCOMES

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Noise-reduction algorithms are known to provide significant improvements in speech perception for cochlear implant recipients in noisy situations. Therefore, all cochlear implant should now include this feature, as the use of such algorithms has been progressively standardized in clinical practice for patient fitting. The goal of the present report is to present the industrial development process of such an algorithm, from software design and implementation, to outcome measures in patients.

A noise-reduction method using one microphone has been developed and optimized to hardware constraints. In order to fit with current coding strategy, noise estimation and reduction were post-processing based. This algorithm integrates noise estimation based on spectral minimum over a time-window, evaluating the steady background noise spectral profile. This profile is finally subtracted spectrally, taking into account the signal information in order to maintain overall loudness of target signal.

This noise-reduction algorithm was tested with 20 cochlear implant recipients over 2 test sessions. Speech perception was assessed through speech in quiet and in noise tests with two noise types: steady-state speech-shaped noise (5dB SNR) and multitalker babble (0, 5 and 10dB SNR). Pure tone thresholds were also assessed. Finally, a questionnaire was used to evaluate sound quality and everyday life listening situations. Subjects were first tested immediately after noise reduction activation, then after one month of use.

Results showed that noise-reduction algorithm did modify neither pure-tone thresholds, nor speech intelligibility in quiet. Moreover, significant improvement in steady noise has been observed. With regards to implementation hypothesis and results, the noise-reduction algorithm can then be clinically validated. Besides, significant speech perception improvement was found in babble noise. Listening quality testing for noise annoyance and overall preference also found significant improvements.

This noise reduction algorithm has shown significant speech perception and quality improvements compared to the standard processing. It is now proposed for clinical practice.

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R49: EFFECT OF TRANSIENT NOISE REDUCTION ON SPEECH INTELLIGIBILITY AND SOUND QUALITY IN CI USERS

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Speech understanding in noisy environments remains a challenge for cochlear implant (CI) users. Single-microphone noise reduction algorithms have been shown to improve CI speech performance, but are typically designed for situations where the noise is relatively stationary. Many environmental noises, however, are characterized by rapid fluctuations in level, such as the sound of clattering dishes or footsteps on a hard surface. Transient noise reduction (TNR) algorithms specifically tailored to such types of noise are already available as options on state-of-the-art hearing aids. This study assessed the effect of one such algorithm on speech intelligibility and sound quality in CI users.

Twelve users of the Advanced Bionics HiRes90k implant took part in an acute study. The clinically-fitted HiRes120 strategy, without noise reduction, was compared to two settings of a commercial hearing aid TNR algorithm, added as a pre-processing stage. Speech reception thresholds (SRTs) in two types of transient noise (repetitive hammering and clinking of spoon against cup) were measured using the material of the Oldenburg sentence test (OlSa). Speech intelligibility at a conversational level in quiet was assessed with the Hochmair-Schulz-Moser (HSM) sentence test. For comparisons regarding subjective sound quality, subjects ranked the three processing options in terms of speech clarity, comfort and overall preference for the signals used during speech testing. Signals were generated on a laptop computer running a simulation of the TNR algorithm and delivered to the subjects' speech processors electrically.

For the stronger of the two TNR settings, OlSa SRTs in noise improved significantly over the baseline strategy by approximately 2dB in hammering noise and 1dB in clinking noise. HSM sentence scores in quiet were at ceiling for all three processing conditions and showed no adverse effect for either TNR setting. Rank differences between the conditions revealed a significant preference in terms of speech clarity in hammering noise for both TNR settings over the baseline condition, and a tendency towards improved clarity and comfort overall. Our results demonstrate that a commercially-established transient noise reduction algorithm for hearing-aid users can improve CI speech perception in at least some fluctuating noise types.

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R50: EVALUATION OF AGING CHARACTERISTICS OF PIEZOELECTRIC ARTIFICIAL BASILAR MEMBRANE

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The sound waves were transmitted to the inner ear through the tympanic membrane and the middle ear. The cochlea of the inner ear has two core mechano-sensory components that are the basilar membrane and the hair cells with the stereocilia. The basilar membrane separates the sound wave mechanically by their frequencies. The hair cells on the basilar membrane generate bioelectric signals as the vibrating basilar membrane. This signal travels to the brain along the auditory nerves

We have developed the artificial basilar membrane (ABM) that mimics the function of the human basilar membrane and the hair cells by using the piezoelectric thin film as a membrane material. This piezoelectric ABM does not require the external energy source and signal processing unit to separate the frequency and generate the electric signal. We use the polyvinylidene fluoride (PVDF) thin film as the piezoelectric ABM. Polymeric piezoelectric materials are apt to age and change the mechanical properties and piezoelectric characteristics.

In order to guarantee the reliability of the frequency selectivity and the generating of electrical signal of ABM, we have to estimate the aging characteristics of ABM. Generally the accelerated aging test procedure has been used in evaluation of the aging properties of polymeric materials. The accelerated aging tests were prepared under several ranges over room temperatures, and these tests were continued 1 to 60 days. It was shown that there was no significant degradation in elastic modulus, tensile strength and piezoelectric property of ABM.

We predict that PVDF artificial basilar membrane can be one of the core components of totally implantable artificial cochlear system.

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R51: SELF-SELECTION OF COCHLEAR IMPLANT SPEECH PROCESSING PARAMETERS

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Optimization of the parameters that control cochlear implant (CI) speech processing algorithms on an individual by individual basis can have a substantial impact on speech recognition. Currently, parameter fitting occurs with the aid of clinicians, and these fittings are routinely updated over time to compensate for changing physiology and adaptation of perception. However, since each re-fitting requires a clinic visit, the frequency of fitting may be limited by time or cost constraints rather than user need. A system that supplements clinical fitting by allowing parameter updates in the home may result in reduced medical costs as well as improved performance and satisfaction since the limitations on the frequency of fitting would be mitigated. The remote delivery of health care services outside of the hospital or clinic (termed 'telehealth') has been demonstrated to be beneficial for a variety of chronic or long-term care needs by decreasing medical costs, increasing access to health care, and increasing the quality of care. Further, several studies have indicated that cochlear implant recipients are able to update their parameters with software-based, rather than clinician-based, guidance. Thus, a home-based fitting system is a realistic option for increasing the benefit that individuals receive from their implant while reducing the burden of medical costs and time requirements.
A software interface has been developed that allows for the modular addition of tasks with which CI users can interact in order to adapt their current speech processing parameters. Currently, the software provides a method of selecting threshold and maximum comfort levels, the loudness growth compression parameter Q, and pulse rate. The software includes multiple speech recognition tasks that can be used to assess performance and satisfaction with new parameter values. Each task has been assessed by CI users during the development process to create an interface that is self-explanatory and easy to use.

Clinical parameter values are compared to self-selected parameter values for three Nucleus CI users. Speech recognition performance is compared between these two parameter sets, and CI users' qualitative assessments of the process of self-selecting parameters will be included.

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R52: EFFICIENT CODING FOR AUDITORY PROSTHESES

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Conventional cochlear implant coding strategies (e.g. ACE, CIS) convey sound information from multiple spectral bands by sampling the temporal envelope of each channel with fixed-rate pulses. We investigated an experimental sound coding strategy, called Fundamental Asynchronous Stimulus Timing (FAST), which employs a sparse, yet temporally precise, representation of temporal envelope. We hypothesized that the FAST strategy would provide better temporal sensitivity to ITD and F0 cues while using significantly less stimulation power than conventional strategies.

Several types of listening experiments were conducted comparing the ACE and FAST strategies in unilateral and bilateral Nucleus cochlear implant subjects. These include 1) loudness comparisons, 2) speech understanding in quiet and in noise, 3) ITD sensitivity of speech stimuli, 4) ITD-based spatial release from masking, and 5) F0 sensitivity. Additionally, battery life during take-home usage was collected to compare power consumption across strategies.

Results from the loudness comparisons suggest that for ACE rates > 720 pps, FAST is as loud, or louder, than ACE with matched T- and C-levels, even though FAST uses fewer overall pulses. Acute speech understanding in quiet was initially comparable across strategies but worse for speech in noise with FAST. After several weeks of take-home experience with FAST, however, speech-in-noise scores recovered to those with ACE. In tests with bilateral subjects, significantly better ITD sensitivity and ITD-based spatial release from masking were observed with FAST. Additionally, subjects reported improved perception of intonation and at least double the battery life with FAST over ACE.

Results suggest that a temporally sparse coding strategy, such as FAST, provides a sufficient representation of speech sounds while allowing for more precise and efficient coding of temporal envelope. This development may lead to better hearing outcomes with auditory prostheses and future opportunities for device miniaturization.

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