Arches in Paris

ARCHES Meeting

25th – 26th November 2019

Programme and Abstracts

École normale supérieure, Paris, France

Welcome

Dear you all,

It is our great pleasure to welcome you at École normale supérieure in Paris for the 2019' session of the ARCHES network.

We thank you warmly for your many proposals for oral and poster presentations on topics as diverse as effects of sensorineural hearing loss, cochlear-implant or hearing-aid processing on auditory localization, informational masking, speech intelligibility in adverse listening conditions, well-being, musical perception or listening effort. European research in experimental and clinical audiology is quite lively and productive, interdisciplinary by nature, and the coming workshop will give a new demonstration.

As every year, these presentations and open discussions will contribute to significantly increase our mutual awareness, to enhance student, postdoc and PI mobility and to set up novel and exciting collaborations between research units across Europe.

We sincerely hope that you will enjoy these two days of discussion, debate and networking in Quartier Latin.

A bientôt!

Christian Lorenzi

ARCHES 2019 team:

Christian LORENZI

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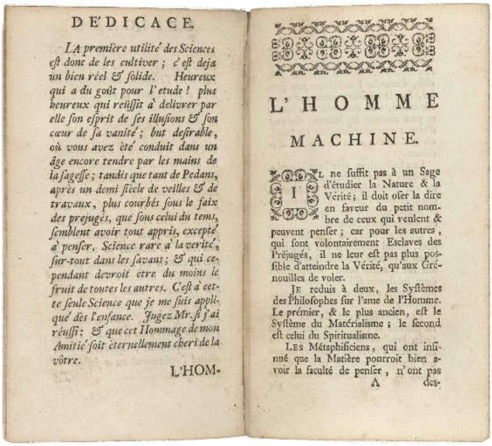

DE'DICACE.

LA première utilité des Sciences eſt donc de les cultiver ; c'eſt deja un bien réel & ſolide. Heureux qui a du goût pour l'etude ! plus heureux qui reüſſit à delivrer par elle ſon eſprit de ſes illuſions & ſon cœur de ſa vanité ; but deſirable, où vous avez èté conduit dans un âge encore tendre par les mains de la ſageſſe ; tandis que tant de Pedans, après un demi ſiécle de veilles & de travaux, plus courbés ſous le faix des prejugés, que ſous celui du tems, ſemblent avoir tout appris, excepté à penſer. Science rare à la verité, ſur-tout dans les ſavans ; & qui cependant devroit etre du moins le fruit de toutes les autres. C'eſt à cette ſeule Science que je me ſuis appliqué dès l'enfance. Jugez Mr. ſi j'ai réüſſi : & que cet Hommage de mon Amitié ſoit èternellement cheri de la vôtre.

L'HOM-

L' HOMME

MACHINE.

IL ne ſuffit pas à un Sage d'étudier la Nature & la Vérité ; il doit oſer la dire en faveur du petit nombre de ceux qui veulent & peuvent penſer ; car pour les autres, qui ſont volontairement Eſclaves des Préjugés, il ne leur eſt pas plus poſſible d'atteindre la Vérité, qu'aux Grénouilles de voler.

JE reduis à deux, les Syſtèmes des Philoſophes ſur l'ame de l'Homme. Le prémier, & le plus ancien, eſt le Syſtème du Matérialisme ; le ſecond eſt celui du Spiritualisme.

LES Métaphiſiciens, qui ont inſinué que la Matière pourroit bien avoir la faculté de penſer, n'ont pas

A des-

The Paris' Bonus:

Current research in hearing science and audiology is guided by computational models of auditory mechanisms. The 'modulation filterbank' model is a perfect example of a successful model of auditory mechanisms that contributed substantially to a better understanding of normal and impaired hearing at the beginning of the XXIth century. Producing such models of mechanisms is at the heart of modern science. René Descartes, a French mathematician and philosopher of the XVIIth century who spent a large part of his life in Netherland and died in Sweden, initiated and paved the way for the mechanical approach of natural phenomena such as auditory perception. In a sense, the Cartesian approach is still alive and it may be considered to be part of the ´conceptual core´ of the European scientific enterprise. The 2019´ session of the ARCHES workshop will offer us an opportunity to contemplate original editions of his books (e.g. *Traité de l'Homme*) at the historical library of Ecole normale supérieure and discuss openly about these fundamental principles of scientific research.

Image credits: Wikipedia & Gallica

Abstracts

Monday talk sessions

Mon 13.10 - Speech and voice perception: an intricate relationship

Etienne Gaudrain, Leanne Nagels, Nawal El Boghdady, Christina Fuller, Rolien Free, Deniz Başkent

CNRS Lyon; University Medical Center Groningen

When two people talk at the same time, it becomes challenging to follow what they say if their voices sound similar. The ability to discriminate voices seems crucial in cocktail party situations. However, the relationship between the ability to detect a small difference between two voices and the benefit one may derive from this voice difference to understand two competing talkers is not straightforward. The acoustic cues associated with voice differences are indeed complex, and intertwined with acoustic cues associated with linguistic information. The ability to detect and exploit voice differences thus results from a cognitive process that slowly develops over childhood. By studying populations with special hearing abilities, like musicians or cochlear implant users, recent studies have shed some light on this intimate relationship between form and content in speech perception. These findings also offer new perspectives for the improvement of hearing to multiple speakers with cochlear implants.

Mon 13.30 - Evaluation of a speech intelligibility model for predicting speech intelligibility in normal hearing and hearing-impaired listeners

Helia Relaño-Iborra[1], Johannes Zaar[1] and Torsten Dau[1]

[1] Hearing Systems Section, Department of Health Technology, Technical University of Denmark, DK-2800 Kgs. Lyngby, Denmark.

A speech intelligibility model is presented based on the Computational Auditory Signal Processing and Perception model (CASP) of Jepsen, Ewert, and Dau (2008) [J. Acoust. Soc. Am. 124(1), 422–438]. The model combines a non-linear auditory-inspired preprocessing with a back end based on the cross-correlation between the clean and the degraded speech representations in the modulation envelope domain. Several speech degradation and speech enhancement algorithms were considered to study the ability of the model to predict data from normal-hearing listeners. Degradations of speech intelligibility due to additive noise, phase-jitter distortion and single-channel noise reduction as well as improved speech intelligibility due to ideal binary mask processing are shown to be successfully accounted for by the model. Furthermore, the speech-based CASP (sCASP) was evaluated as a predictor of speech intelligibility for hearing-impaired listeners. Speech reception thresholds obtained in two different studies (Christiansen and Dau, 2012; Johannesen et al., 2016) with different speech materials and masking noises were considered. Experiment 1 focused on predicting aided speech perception in the presence of stationary noise and experiment 2 considered speech perception of (unaided) listeners in the presence of stationary and fluctuating noises. The model was evaluated in terms of its predictive power of the listener's average data and as a predictor of individual results across listeners with varying degrees of hearing loss. Overall, the model accounted well for trends observed at a group level, whereas significant correlations between the measured and the predicted performance across the individual listeners were only observed in one of the considered data sets

Mon 13.50 - Speech intelligibility and context effects

Jelmer van Schoonhoven[1], Koenraad S. Rhebergen[2], Wouter A. Dreschler[1]

[1]Department of Clinical and Experimental Audiology, Amsterdam University Medical Centre

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The Speech Transmission Index (STI) can be used to predict speech intelligibility in stationary noise and reverberation Houtgast & Steeneken (1973). STI measurements are not validated for fluctuating background noise. We presented an extended version of the STI (van Schoonhoven et al., 2019) that makes use of the instantaneous STI values in order to improve accuracy in fluctuating noises. Predictions improved, but for low modulation rates (<8 Hz) the model still deviates from actual speech intelligibility data. One of the hypotheses is that context is a limiting factor. Several context models are available in the literature, most notably by Boothroyd & Nittrouer (1988) and by Bronkhorst et al. (1993). The goal of the current research is to model the effect of context in order to improve speech intelligibility prediction in fluctuating noise with slow modulations. Monaural speech intelligibility experiments in normal hearing subjects were conducted using monosyllabic sense and nonsense words presented over headphones. Speech shaped stationary noise and interrupted noise (0.5-16 Hz) were used as background noise. Also, the speech was interrupted by silent periods using the same rates. These speech intelligibility data will be presented, together with the steps to implement context effects into the model and to improve model accuracy.

Mon 14.40 - Localization with and without hearing devices in plausible environments.

Laurent S. R. Simon[1], Andrea Kegel[1], Hannes Wüthrich[2], Norbert Dillier[1]

[1] Laboratory of Experimental Audiology, ENT Department, University of Zurich, Switzerland

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Localization with hearing-devices in complex environments is still not well studied, particularly in terms of perception of elevation. Two localization experiments have therefore been conducted and will be compared. In both experiments, users had to localize a target sound that was coming from one of 32 directions simulated using Higher Order Ambisonics, with or without hearing aids. In the first experiment, normal-hearing persons had to localize bursts of anechoic pink noise. In the second experiment, hearing-impaired subjects had to localize a speech source in either a restaurant or in a street environment. The speech source was convolved with HOA room-impulse responses recorded in each environment.

Results showed that the use of hearing aids increases the time required to localize sounds for the normal-hearing and half of the hearing-impaired participants. They also showed that hearing-impaired participants had no perception of elevation, with or without hearing aids, and that the type of hearing aid beamformers had a different effect on reaction time depending on the participant's hearing loss.

Acknowledgements: This work was funded by Innosuisse Swiss Innovation Agency.

Mon 15.00 - Evaluation and optimization of hearing devices in 3D complex environments

Ruksana Giurda[1,2], Laurent Simon[1], Norbert Dillier[1]

[1] Laboratory of Experimental Audiology, University of Zürich

[2] Department of Information Technology and Electrical Engineering, ETH, Zürich

Sound localization can be a challenging task in adverse acoustic environments. Hearing impaired listeners, even with hearing devices, show lower performances in localization tasks with respect to normal hearing people [1]. Currently, hearing aids use sound classifier to select the best settings accordingly with the acoustic environment detected. These classifiers, however, do not use any spatial information. The goal of this project is to design and implement a localization algorithm for bilateral hearing aids fitting which can improve the performance of the existing sound classifier by using the location of the detected sound sources. An auditory map consisting of 6 regions, three in the front hemisphere (front, left, right) and three in the back hemisphere is virtually created around the hearing aid user. The map is constantly updated over time with the output of the localization algorithm which detects potential sound sources in each one of the 6 areas. Based on the detected sound sources, and their location, a beam-former is used to focus the acoustic attention in specific directions. The sound classifier is then run on the beamformed signals. In order to have a robust localization even in presence of reverberation, a blind DRR estimator based on acoustic onsets has been developed. The estimator is based on efficient linear regression with acoustic onsets as regressors. The model works by using the acoustic inputs of a single microphone, however, estimation from left and right devices can be efficiently combined in the case of binaural fitting. In the case of head movements, the auditory map needs to be updated accordingly. Therefore, the information coming from accelerometer sensors will be combined with the localization algorithm. Preliminary tests have been already conducted to design a robust head-movement detector which, in case of head movement, can also estimate the rotation in the azimuth plane.

[1] Terese Finitzo-Hieber and Tom W Tillman, "Room acoustics effects on monosyllabic word discrimination ability for normal and hearing-impaired children," Journal of speech and hearing research, vol. 21, no. 3, pp. 440–458, 1978.

Acknowledgement: Funded by Innosuisse. Project n. 34750.1 IP-LS.

Mon 16.20 - Inter-individual differences above the detection threshold

Daniel Pressnitzer [1]

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Normal hearing listeners can display a wide range of inter-individual differences in their supra-threshold auditory percepts. This is vividly illustrated by auditory illusions, as for instance in the recent Laurel/Yanny case, for which a same speech utterance is reported as completely different words by different listeners. Such a variability may also have practical implications for impairments that are not properly diagnosed by standard audiological tools. In this talk, I will briefly survey an ongoing line of work showing that auditory experience continuously alters auditory processing, which could be one cause for inter-individual variability. I will also introduce preliminary data suggesting that informational masking of a tone above detection threshold in silence is frequency-dependent, in an idiosyncratic manner, in normal hearing and hearing-impaired listeners.

Mon 16.40 - Factors influencing need for recovery in employees with hearing loss: a cross-sectional study of health administrative data.

Hanneke E.M. van der Hoek-Snieders, Monique Boymans, Bas Sorgdrager, Wouter A. Dreschler

Department of Clinical and Experimental Audiology, Amsterdam University Medical Centre

Objective: Need for recovery is a predictor of work stress, health problems, and sick leave, but its underlying factors are not yet well understood. We aimed to identify disease-specific, work-related, and personal factors influencing need for recovery in hearing impaired employees.

Methods: We retrospectively identified hearing impaired employees (N = 294) that were referred to the Amsterdam University Medical Center between 2004 and 2019. Routinely health care data was used, including a hearing survey and hearing assessments. A directed acyclic graph was constructed, revealing the underlying structure of factors influencing need for recovery as well as the minimal set of factors needed for multiple regression analysis.

Results: The directed acyclic graph showed four variables directly influencing need for recovery explaining 46.1% of the variance, specifically: need for change at work ($B = 19.01$, $p < .001$), self-perceived listening effort ($B = 1.84$, $p < .001$), personal adaptations scale score ($B = -.34$, $p < .001$), and having a moderate/poor health condition ($B = 20.06$, $p < .001$). Although degree of hearing loss was associated with self-perceived listening effort, there was no significant association between degree of hearing loss and need for recovery.

Conclusions: The results suggest that the way employees perceive their hearing loss and how they cope with it directly influence need for recovery, rather than their measured degree of hearing loss. Additionally, general health condition was found to be an independent factor for need for recovery. The results should be confirmed by future, longitudinal research.

Mon 17.00 - Longitudinal evaluation of hearing status in professional musicians

Dorien Verschuren[1], Wouter Dreschler[1]

[1] Department of clinical and experimental audiology, Amsterdam UMC location AMC, The Netherlands

For professional musicians their hearing is their most important instrument. And yet, they have an increased risk of developing hearing problems because their jobs expose them to excessively high sound levels. . While hearing problems are diverse and vary greatly from person to person, research and clinical evaluation mostly focuses on shifts in audiometric thresholds. For musicians, however, problems such as tinnitus or diplacusis are just as pressing, while being harder to treat. In more detail, diplacusis is the audiological phenomenon that a pure-tone results in different pitch percepts in each ear. To musicians, it is important to be aware in case of strong diplacusis, as this phenomenon can affect their ability to tune, and even play, their instruments or voice. The present study therefore aims to increase our knowledge of the long term effects on hearing status due to prolonged exposure to loud music.

To this end, 42 out of 245 Dutch orchestra musicians who participated in the previous study (Jansen et al., 2009) and were still employed, were examined for a range of hearing complaints including shifts in audiometric thresholds, diplacusis, tinnitus, and speech intelligibility in noise. Also, they were asked about subjective hearing symptoms. The results were then compared to baseline measurements taken 14 years earlier, to provide insight into the hazards of professional musicianship to hearing (problems). Results will be shown at ARCHES.

Tuesday talk session

Tue 09.00 - Neural generators of auditory temporal processing across different age cohorts

Ehsan Darestani Farahani [1], Jan Wouters [1], Astrid van Wieringen [1]

[1]*Research Group Experimental ORL, Department of Neurosciences, KU Leuven - University of Leuven, Belgium*

Objectives: Many middle-aged and older people, even with normal audiometric threshold, experience difficulties in speech understanding, especially in noisy environments or when multiple speakers are talking. This problem can be attributed to changes in central temporal processing. To gain a better understanding of these changes, it is of great (clinical) interest to investigate the neural generators of auditory temporal processing across age.

Methods: To investigate temporal envelope processing we used auditory steady-state responses (ASSRs), auditory evoked responses which reflect neural synchrony. We also presented a novel extension to minimum-norm imaging (MNI) which facilitates ASSR source reconstruction. The neural sources were reconstructed in young, middle-aged and older persons with normal audiometric thresholds. For each neural source, the ASSR amplitude and phase coherence were obtained to investigate the degree of synchronized activity and the phase-locking ability, respectively. This was done for low and high rate modulation frequencies, presented to both the left and right ears.

Conclusions: First, we demonstrate that the proposed MNI approach is successful in reconstructing cortical and sub-cortical sources. Furthermore, results indicate that (1) in older adults compared to younger ones, the response amplitudes are increased, while the phase-locking does not change when the modulations are relatively slow (< 50 Hz); (2) that phase-locking of faster modulations (> 50 Hz) may be compromised, together with reduced response amplitudes.

Acknowledgment: Our special thanks go to Dr. Tine Goossens for sharing the data used in this work. This work was supported by the Research Council, KU Leuven through project OT/12/98 and by the Research Foundation Flanders through FWO-project ZKC9024 and a TBM-FWO grant from the Research Foundation-Flanders (T002216N).

Tue 09.20 - Age-dependent changes in frequency-following responses as a potential marker of cochlear synaptopathy in humans

Jonatan Märcher-Rørsted [1], Jens Hjortkjær [1], Gerard Encina-Llamas [1], Torsten Dau [1]

[1]Hearing Systems Section, Department of Health Technology, Technical University of Denmark

Cochlear synaptopathy occurs as a consequence of noise exposure and aging but diagnostic measures in humans are missing. With synaptopathy, a reduction of the number of auditory nerve fibers may degrade the processing of fine temporal cues relying on synchronous activity of many nerve fibers. The frequency following response (FFR) is considered to reflect synchronous neural activity phase locked to the temporal fine structure of the stimulus. At higher stimulus levels, due to the spread of neural excitation across frequency, the FFR represents a fairly broadband response dominated by the synchronized neural activity stemming from more basal ("off-frequency") nerve fibers. A degraded neuronal synchrony due to loss of nerve fibers may lead to a reduced FFR, even for stimulus frequencies where no sensitivity loss is found. Here we investigate age-related changes in FFRs and EFRs to amplitude-modulated tones at 706 Hz and 326 Hz, presented at a sound pressure level of 85 dB nHL. Furthermore, FFRs to frequency sweeps from 0.2 to 1.2 kHz were measured to explore the upper limit of the FFR existence region in young and older listeners. Additional potential measures of synaptopathy such as the middle-ear muscle reflex, high-frequency audiometry and early components of the auditory brainstem responses were also evaluated. The results demonstrate a reduction in the higher frequencies of the FFR response in aging listeners. A reduction of synchronous activity at higher frequencies with synaptopathy were also consistent with simulations using a computational model of the auditory-nerve.

Tue 09.40 - Objective measures of temporal modulation encoding in the auditory pathway and its relevance for cochlear implant fitting.

Robin Gransier [1] & Jan Wouters [1]

[1]KU Leuven, Department of Neurosciences, ExpORL, Leuven, Belgium

Temporal envelope modulations (TEMs) are the primary cues that cochlear implant (CI) users utilize to perceive speech. The ability of the electrically stimulated neural ensembles to encode TEMs is important for speech perception with a CI, especially in adverse listening situations. Behavioral assessments of TEM encoding (i.e. modulation detection thresholds) have shown that the ability to encode TEMs varies across the neural ensembles allocated across the CI electrode array and that this variability is related to the speech perception outcome with a CI. It is, however, poorly understood how this channel-depended TEM encoding is represented at the different stages of the auditory pathway. Two potential noninvasive electrophysiological measures to assess neural TEM encoding are the auditory steady-state response (ASSR) and the acoustic change complex (ACC). The ASSR is a phase-locked response to the temporal structure of the stimulus that originates, depending on the modulation frequency, from different regions of the auditory

pathway, whereas the ACC originates from the auditory cortex and can be elicited by a change from a constant to a modulated stimulus. Measuring these electrophysiological responses is, however, challenging due to the CI-stimulation artifacts that contaminate the EEG recordings, especially for clinically relevant stimulation parameters.

We will present an overview of a number of recent studies originating from our lab where we investigate how the neural TEM encoding can objectively be assessed in CI users by means of the ASSR and the ACC, and how the outcome of these measures, especially that of the ASSR, can be used to gain insight in the speech processing ability of the CI recipient. More specifically, we look at how modulation depth affects both electrophysiological measures and how the variability in TEM encoding across the array, as assessed with the ASSR can be used to predict speech perception in noise with a CI. Furthermore, we will discuss the potential clinical relevance of these measures for CI fitting.

Tue 10.00 - Immediate effects of acoustic trauma on neural envelope coding in the inferior colliculus

Amarins N. Heeringa [1,2,3] & Pim van Dijk [1,2]

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Despite deteriorated sensitivity and difficulties with understanding speech in noisy environments, patients with sensorineural hearing loss have normal to better-than-normal envelope detection thresholds. Correspondingly, phase locking to the sound envelope is improved at the level of the auditory nerve four weeks after acoustic trauma. The underlying mechanisms for this phenomenon are, however, not fully understood. Here, we studied the immediate effects of acoustic trauma on temporal and rate coding of the sound envelope by inferior colliculus neurons.

Neural activity in response to amplitude-modulated noise (modulation frequencies = 8 – 1024 Hz, octave-based steps; modulation depth = 50%; stimulus level = 70 dB SPL) was recorded from the inferior colliculus of the guinea pig, before and immediately after a 1-hr 11-kHz, 124 dB SPL, bilateral acoustic trauma.

Units with a characteristic frequency (CF) below the trauma frequency (< 11 kHz) had an average immediate threshold shift of 14 dB. Nevertheless, these units had significantly increased response gains, a measure for phase locking to the sound envelope, especially at low modulation frequencies (≤128 Hz). High-CF units (CF > 11 kHz), which had a larger acoustic trauma-induced threshold shift (> 50 dB), remained responsive to the amplitude-modulated noise but with decreased response gains. Furthermore, driven firing rates decreased, especially at high modulation frequencies for high-CF units.

The observed changes occurred immediately following trauma and were thus a result of the immediate trauma-induced damage to the auditory system. The enhanced temporal coding by low-CF units may underlie the normal to better-than-normal envelope detection thresholds in patients with sensorineural hearing loss. Moreover, if also present in human subjects, reduced response gains in high-frequency units could disrupt coding of consonants and consequently impair speech understanding in noisy environments.

Tue 11.50 - The 'missing 6 dB' revisited: Loudness mismatch between headphone and loudspeaker presentation

Florian Denk [1], Josep Llorca-Bofí [2], Michael Kohnen [2], Michael Vorländer [2], Birger Kollmeier[1]

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[2] Institute of Technical Acoustics, RWTH Aachen University, Germany

Since the 1930s, generations of researchers observed a mismatch between headphone and loudspeaker presentation: The sound pressure level at the eardrum generated by a headphone has to be about 6 dB higher compared toa free-field source to elicit the same perceived loudness. While it has been shown that thiseffect vanishes if the same waveforms are generated at the eardrum in a blind comparison, the origin of the mismatch is still unclear. We present new data on the issue that systematically characterizes this mismatch under variation of the stimulus frequency, presentation room, headphone type, and binaural parameters of the headphone presentation. Subjects adjusted the playback level of a headphone presentation to equal loudness as loudspeaker presentation; the levels at the eardrum were determined through appropriate transfer function measurements. Identical experiments were conducted at Oldenburg and Aachen with 40 normal-hearing subjects, including 14 that passed through both sites. Our data verifies a mismatch between loudspeaker and diotic headphone presentation, especially at low frequencies. This mismatch depends on the room acoustics of the loudspeaker presentation, and strongly onthe interaural correlation in both presentation modes.

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Tue 12.10 - Loudness: From cochlear implants to complex audio-visual scenes

Stephan D. Ewert [1], Hongmei Hu [1], Iko Pieper [1], Thomas Biberger [1], and Stefan Fichna [1]

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Loudness is a perceptual attribute that can intuitively be assigned to any sound without training. Loudness is highly relevant in many areas ranging from hearing aid fitting and satisfaction, noise and soundscapes, music and sound mastering to fundamental psychoacoustics as a measure of supra-threshold sound intensity. Recent studies on loudness perception of binaural broadband signals in hearing impaired listeners found large individual differences, suggesting the use of such signals in hearing aid fitting and triggering further research in the direction of cochlear implants (CI), individual loudness models, and loudness perception in realistic scenarios. In this talk, loudness perception and the application of loudness measures are considered in three connected areas i) diagnostics and fitting of bilateral cochlear implant and single-sided deafness (SSD) patients, ii) computational loudness models for individual normal-hearing (NH) and hearing-impaired (HI) listeners, and iii) loudness in life-like audio-visual scenes. First, spectral and binaural loudness summation was investigated in NH listeners, bilateral CI (biCI) users, and SSD CI (ssdCI) users with normal hearing in the unaided ear. Categorical loudness scaling was performed for an equal categorical loudness noise (ECLN) consisting of the sum of six spectrally separated third-octave noises at equal loudness. Simultaneous electrical and acoustical stimulation (or either) were used. Results showed a higher (spectral) loudness summation of equally loud narrowband signals in CI compared to NH, resulting in unbalanced bilateral loudness of equally loud narrowband signals for ssdCI users. The results suggest that the binaural ECLN applied here could improve fitting and binaurally balanced loudness perception at least for ssdCI users. Second, a monaural and binaural, physiologically motivated loudness model was suggested, which can be fitted to account for individual loudness perception of NH and HI for narrowband, broadband and binaural stimuli. Existing loudness models cannot be adapted to account for individual data. The current model might be helpful for individual aided performance prediction and is suited to distinguish the effect of outer and inner hair cell loss from additional individual parameters such as a hypothesized central gain. Third, in daily life, auditory perception is typically accompanied by the visual impression of our surroundings. To enable auditory perception experiments in simulated life-like scenarios, we connect realistic, immersive visual renderings of a state-of-the-art computer game engine with our freely-available tools for psychoacoustic experiments in MATLAB (AFC,www.aforcedchoice.com) and perceptually optimized and evaluated room acoustics simulation (RAZR,www.razrengine.com). An experimental setup for loudness perception of single auditory objects in the context of other sound sources and reverberation is shown.

Poster sessions

Classifying match/mismatch between EEG and stimulus using deep learning

Bernd Accou [1,2], Mohammad Jalilpour Monesi [1,2], Hugo Van hamme [1], Tom Francart [2]

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[2] *KU Leuven, Dept. Neurosciences, ExpORL*

Recent advances in signal processing and neuroscience have shown that the envelope of a speech stimulus can be decoded from the EEG using a linear model. Given the complex and non-linear nature of the brain, we investigated more complex deep learning models. Instead of tackling the difficult regression problem, we reformulated the problem as a classification problem: which of two given envelopes best matches the EEG.
We evaluated the paradigm using 10-second segments of EEG and (mis)matched envelope from our dataset, in which 90 subjects listened to natural running speech. The mismatched envelope segment was retrieved from the same dataset, but 1 second before or after the EEG and matched envelope segment. We will present a comparison between a baseline linear model and various convolutional neural networks.

Evaluating the Contribution of Cross-Channel Modulation Interference to Listening Effort of Cochlear-Implant Users during Speech-on-Speech Recognition

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Cochlear implants (CI) provide mainly amplitude modulation (AM) cues. AM processing is subject to cross-channel masking or interference effects, also known as Modulation Detection Interference (MDI), whereby detection or discrimination of a target AM is massively affected by AM of a spectrally-distant distractor. However, the contribution of MDI to the listening effort of CI users in everyday adverse listening conditions remains largely unaddressed. In this study, we intend to measure listening effort of both normal-

hearing and CI listeners while they perform a speech-recognition task under different noise conditions, as well as during an AM detection task under different distractor conditions, using the pupil dilation response to auditory stimuli as a proxy for listening effort. We expect that listening effort increase for speech-on-speech compared to speech-in-quiet recognition will be correlated to the distractor modulation effects on AM detection performance and on associated listening effort in CI users who massively rely on AM cues, but less so or not at all in normal-hearing subject who have access to other (e.g. spectral temporal fine structure) cues for speech recognition.

Spatiotemporal Dynamics of Auditory Attention Allocation

Bernhard Eurich [1], Manfred Mauermann [1] and Birger Kollmeier [1]

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In complex acoustic scenes, normal hearing listeners are able to focus their attention to one single talker while suppressing the neural processing of others („Cocktail Party Effect"). This perceptual separation is based on a dynamic interplay between saliency-driven bottom up cues and deliberate selective attention on auditory objects ("top down"). It might be assumed that attention is switching between time slices or chunks of distinctive duration. For example, it has been repeatedly shown in EEG measurements that neural dynamics entrain to the syllable structure of speech, associated with a neural strategy of parsing speech into short chunks while dynamically attending to time points of predicted acoustic events (temporal predictability). This, however, appears to happen only in a narrow frequency band of everyday speech syllable rates (theta band). In order to characterize these chunks, it is now hypothesized that this dependency on syllable rate and temporal predictability should also influence detection of speech segments and cognitive load. In a psychoacoustic experiment in the format of a behavioral oddball task, accuracy rates and reaction times were measured for detection of deviant syllables in a target stream while ignoring a running speech stream. This was done for syllable rates within the theta range as well as below it. Very slow inter-onset intervals turned out to result in significantly longer reaction times and a higher number of errors in response behavior. Little irregularities in the inter-onset intervals, however, did not produce significant effects. This is interpreted as an impairment of selective attention and at the same time an increase of cognitive load when attending a target speech signal with a syllable rate below theta. This results underline the special role of the theta range on attention dynamics and the need to consider the dependency of presumed time windows on the temporal structure of the attended sound in order to characterize dynamic auditory attention.

Exploring ramped pulse shapes for cochlear implants in an animal model

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A limiting factor in cochlear implants (CI) is the spread of excitation when an electrode is stimulated. In current CI devices, the electrical pulse has a rectangular (Rec) shape. However, recent modelling and in vitro studies suggest that a pulse shape with a ramped slope might be more beneficial because it better matches ion channel dynamics of spiral ganglion cells. Here, we explore biophysically-inspired electrical pulse shapes in vivo.

First, we tested if ramped pulses are more efficient than Rec pulses at the level of the brainstem. Deaf-ened mice were implanted with a 4-channel array and eABR were recorded in response to Rec and three ramped pulses: rampUP (increasing slope), rampDOWN (decreasing slope), or rampLONG (increasing slope over both phases). Less charge, but higher current level amplitude, was needed to evoke respons-es of similar wave II amplitude with ramped shapes compared to Rec shapes (n = 12). The most charge-efficient pulse shape had an increasing ramp over both phases. We also found that a decreasing ramp produced shorter latencies than rectangular pulses. Finally, we demonstrate that the lower threshold could partially be modelled by a simple lowpass filter model, suggesting that simple filter characteristics might contribute to the benefits of ramped pulses for CI-stimulation.

In summary, our study presents the first physiological data on CI-stimulation with ramped pulse shapes in a mouse model. By reducing charge consumption ramped pulses have the potential to produce more battery-efficient CIs and may open new perspectives for the future design of other efficient neural im-plants. The next step is to test if ramped pulses produce less spread of excitation at the level of single neurons. Using multiunit electrode arrays (32-channel), we will record activity of single neurons in inferior colliculus in response to pulse shapes with a pedestal ramp. In these, the pedestal of a given current amplitude drives spiral ganglion neurons to subthreshold and the steepness of the ramp is then used to control the evoked firing rate. Pilot data show that less charge is needed to evoke spike activity of similar firing rate with ramped shapes compared to rectangular shapes (n=1), supporting the eABR results. Future analysis will look into the spread of excitation and temporal jitter with non-Rec pulses.

Objective measures of listening effort in cochlear-implant users

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Cochlear-implant (CI) users report listening to be an effortful task made worse by noisy everyday environments. The ongoing demand for increased mental exertion has negative consequences for communication, participation and potentially long-term cognitive health. Reliable objective measures of listening effort, suitable for CI users, are needed to aid the development and evaluation of effective interventions. This study aims to quantify listening effort in the implanted population by combining two simultaneous physiological measures: brain activity and pupil size. These measures are recorded using fNIRS-based brain imaging and pupillometry techniques, while a group of 24 CI users and 24 normally-hearing controls listen to sentences masked by different levels of background noise. Additional behavioural data and self-reported measures of listening effort are also obtained. Based on previous studies of effortful listening to spectrally degraded speech (e.g. Wijayasiri et al. 2017; Winn et al. 2015), we hypothesize that i) CI users will show increased activation in frontal brain regions, larger pupil diameter and longer response time compared to their normally-hearing peers, reflecting greater listening effort even at highly favourable signal-to-noise ratios; and ii) there will be a correlation between physiological and self-reported measures of listening effort. The results of this study will facilitate an accurate quantification of the cognitive demands of listening through a CI. New insights about the cortical correlates of effortful listening could also inform improvements in CI technology.

Pramudi Wijayasiri, Douglas E.H. Hartley, and Ian M. Wiggins, 'Brain Activity Underlying the Recovery of Meaning from Degraded Speech: A Functional near-Infrared Spectroscopy (fNIRS) Study', Hearing Research 351 (August 2017): 55–67, https://doi.org/10.1016/j.heares.2017.05.010

Matthew B. Winn, Jan R. Edwards, and Ruth Y. Litovsky, 'The Impact of Auditory Spectral Resolution on Listening Effort Revealed by Pupil Dilation': Ear and Hearing 36, no. 4 (2015): e153–65. https://doi.org/10.1097/AUD.0000000000000145.

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Speech in noise perception in childhood: Role of modulation filtering and processing efficiency

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The present study explored the relationship between the capacity to detect amplitude modulation (AM) and speech-in-noise (SIN) identification during childhood. Auditory models suggest that AM detection is not only constrained by the filtering properties of sensory mechanisms in the modulation domain, but also by "processing efficiency", the ability to make optimal use of the available sensory information. Behavioral tasks were designed to assess the development of modulation filtering and processing efficiency of AM cues and its relationship with SIN between 6 and 8 years of age.

Eighty-two children first completed a 2-alternative-forced choice task (AFC) using an adaptive procedure estimating AM detection thresholds for an 8-Hz sinusoidal AM. In this task, the AM carrier was varied in 2 conditions to assess: i) AM sensitivity using a 500-Hz sine tone (No Masking), and ii) AM masking using a 4-Hz wide narrowband noise centered at 500 Hz with small envelope fluctuations (Masking). Second, a "double-pass technique" evaluated the consistency of children's responses for AM detection using a constant-stimuli procedure. Then, AM detection performance in the Masking condition was measured at threshold for 200 trials repeated twice (2 passes) using a 2-AFC task. Percentage of Correct AM detection in each pass (PC) and Percentage of Agreement between the 2 passes (PA) were used to estimate within-listener consistency, a proxy of AM processing efficiency related to internal noise. Finally, children completed an XAB adaptive task measuring consonant identification thresholds in noise using fricatives and stops contrasting over three phonetic features (voicing, place, and manner). Additionally, children completed two standardized tests assessing receptive vocabulary and non-verbal reasoning.

Results showed that AM detection thresholds obtained with both carriers did not significantly improve from 6 to 8 years ($ps > .37$) and all children were similarly affected by AM masking ($p < .001$). When children were tested at threshold, both PC and PA increased with age ($ps = .03$). Thus, AM filtering is not affected by age, but aspects of processing efficiency are. Regarding SIN, thresholds significantly decreased with age ($p = .02$) and were affected by phonetic feature ($p < .001$). Backward regression analyses showed that AM masking associated with vocabulary scores significantly predicted data for Manner (8.9%), PC and PA predicted to a small extent SIN data for Voicing (adjusted $R2 = 5.8\%$) and vocabulary predicted data for Place (5.1%). Overall, processing efficiency, modulation filtering and linguistic level determine SIN identification in childhood.

Auditory Profiling and Profile-based Hearing-aid Processing Strategies: Towards Precision Audiology for Hearing Rehabilitation

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Currently, the clinical characterization of hearing deficits for hearing-aid fitting is based on the pure-tone audiogram only. Implicitly, this assumes that the audiogram can predict performance on complex, supra-threshold tasks. Sanchez-Lopez et al. (Trends in Hearing, Vol. 22, 2018) hypothesized that the hearing deficits of a given listener, both at threshold and supra-threshold levels, result from two independent types of auditory distortion. In their study, a data-driven method for classifying the listeners into four auditory profiles, which differed in terms of their degree of auditory distortions, was proposed and validated. Here, a heterogeneous group of listeners was tested across three locations using a test-battery designed to tap into different aspects of hearing, including speech perception in quiet and noise, loudness perception, binaural processing abilities, and spectro-temporal resolution. The collected data were analyzed using the analysis developed by Sanchez-Lopez et al. (2018), which yielded four clinically relevant patient subpopulations. In the same way that stratified medicine applies specific therapies to specific patient populations, a profile-based hearing-aid fitting strategy was proposed as a form of "precision audiology". In the present study, stratified hearing solutions were tested with listeners belonging to the four proposed auditory profiles. Using a hearing-aid simulator, the listeners' subjective preference for the proposed hearing-aid processing strategies was assessed in various realistic sound scenarios. The results suggested that the different auditory profiles can be associated with different preferences for specific hearing-aid parameters. Moreover, profile-based hearing-aid fitting may be extended to new paradigms of hearing loss compensation and advanced signal processing.

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Probing spectrotemporal modulation processing to better understand supra-threshold hearing deficits

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A largely unresolved problem in auditory sciences concerns the large heterogeneity found among individuals for understanding speech in noisy environments. Although recent studies suggest that suprathreshold auditory mechanisms play a prominent role in these interindividual differences, a precise view of where and how distortions arise along the auditory processing hierarchy is lacking. Spectrotemporal modulations (STMs) provide a unified basis to probe suprathreshold auditory processes as they are recruited with real speech signals. Here, we introduce a novel methodological framework to characterize STM processing combining psychophysical reverse-correlation with computational modeling. This framework was applied to a detection task in normal-hearing (NH) and hearing-impaired (HI) individuals, who were asked to detect a specific STM target embedded in modulation noise. The derived "perceptual filters", which provide an image of how listeners extract the STM target in the presence of other masking STMs, were different between NH and HI listeners, and were also more variable among HI listeners despite comparable losses in audibility. On average, we found that both groups exhibit non-directional band-pass filtering characteristics in the STM space, but that this filtering is biased toward temporal modulations in the HI group. Simulations derived from a simplified version of the modulation filter-bank (MFB) could accurately reproduce the average perceptual filters of NH listeners, and a simple increase in cochlear filters' bandwidths was sufficient to account for the overall shift toward temporal modulations observed in HI individuals. These results indicate that STM processing primarily relies on frequency selectivity and envelope-based mechanisms, but show that additional central mechanisms should be considered to capture interindividual variability within each group. Overall, this joined experimental-modeling approach opens new perspectives to disentangle the different sources of impairment underlying STM processing and therefore contribute to further understand why individuals differ in their suprathreshold hearing abilities.

Simultaneous intra- and extracochlear electrocochleography during cochlear implantation

Leanne Sijgers, Julian Grosse, Flurin Pfiffner, Norbert Dillier, Christof Röösli, Alexander Huber and Adrian Dalbert

Background: Electrocochleography (ECochG) is a promising method to objectively measure changes in cochlear function during cochlear implantation with the aim of reducing surgical trauma. It is hypothesized that a reduction in physiological signals from the cochlea in response to sound during cochlear implant (CI) insertion indicates acute trauma. Different research groups are investigating ECochG signals recorded either from a fixed position near to the cochlea (extracochlear ECochG) or from the tip of the CI electrode array (intracochlear ECochG), and relating these signals to the degree of post-operative hearing loss. However, as the CI electrode array is moving during the recordings, amplitude and phase changes in intracochlear ECochG signals may be caused by variations in the contribution of signal generators along the cochlea to the recorded signal, rather than reflecting cochlear trauma. The aim of this study was therefore to validate intracochlear ECochG recordings with simultaneously recorded extracochlear ECochG recordings.

Methods: Simultaneous extracochlear and intracochlear ECochG signals in response to 500 Hz tone bursts were measured at multiple stages during CI electrode insertion. The sound intensity used varied between 90 dBSPL and 130 dBSPL across patients, but was constant for all recordings during one surgery. The insertion was paused during the measurements and the CI insertion depth during each recording was marked, such that the location of the intracochlear recording electrode could be estimated. The extracochlear recording electrode was placed on the promontory. ECochG changes were correlated with hearing preservation 4 weeks after surgery. Phase and amplitude changes in intracochlear signals were compared with simultaneously recorded extracochlear signals.

Results: A simultaneous amplitude drop of intra- and extracochlear ECochG signals during insertion is associated with a loss of residual hearing. Amplitude drops in intracochlear ECochG recordings can be observed without associated amplitude changes in extracochlear recordings. In some cases, these intracochlear signal drops are accompanied by phase changes, and the signal amplitude may or may not recover at the end of insertion. Phase changes in intracochlear signals can be abrupt or slowly progressing and can also be observed without associated amplitude drops.

Conclusions: Simultaneous amplitude drops in intra- and extracochlear ECochG recordings during insertion of a CI electrode array seem to be associated with cochlear trauma, relevant for hearing preservation. The movement of the recording electrode past different signal generators probably causes phase changes and in some cases amplitude changes in intracochlear ECochG recordings, making interpretation of intracochlear amplitude changes with respect to cochlear trauma difficult.

Comparing speech intelligibility and the role of acoustic-phonetic cues across languages and talkers

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Multilingualism and comparability across languages in the field of audiology is gaining more and more importance. The challenge in investigating the influence of different languages on speech communication and one of the basic measures in audiology – speech intelligibility – is to disentangle language-specific and talker-specific effects. This study compared speech intelligibility in three different languages, namely American English, Spanish and German, including balanced bilingual as well as monolingual talkers in each language. Matrix sentences were used as speech material, which was uttered by four bilingual talkers of German/Spanish, Spanish/English and English/German, as well as by four monolingual talkers of each language. The speech material of matrix sentence tests offers high comparability across languages because of their very similar and controlled construction criteria, their low linguistic diversity and phonetic balance. Speech intelligibility functions were determined for each talker and language in a standard stationary speech-shaped noise (ICRA1) with normally hearing, native listeners. Various acoustic-phonetic parameters, reported in the literature to affect speech intelligibility (including speaking rate, vowel space area, and energy in the mid-frequency region), were determined for each individual talker, in each language. As reported in a previous study, noise had a higher impact on Spanish speech than on German speech. American English was found to be the best intelligible language of the three. Variability across talkers was similar in each of the languages, and larger than the average differences between languages. In each language, the energy in the mid-frequency region of the long-term spectrum was the most important factor influencing intelligibility of a talker.

Explicit and implicit access to ambiguity in frequency shifts

Jackson Graves, Paul Egré, Vincent de Gardelle, Daniel Pressnitzer

In order to succeed in everyday auditory environments, listeners must often distinguish between multiple plausible interpretations of an ambiguous signal. Generally listeners are aware of this ambiguity, but certain classes of ambiguous stimulus seem to inhibit access to ambiguity. We used a well-known ambiguous stimulus consisting of two Shepard tones containing octave-spaced components presented one after the other. We measured listeners' ability to access ambiguity in this stimulus, both explicitly through behavioral responses and implicitly through pupil dilation. In two experiments, listeners judged the direction of the change between the two tones, while pupil diameter was continuously measured. In experiment 1, we varied the degree of ambiguity in the stimulus by varying the size of the interval between tones. In experiment 2, in order to ensure that the observed effects were not due to interval size itself, we compared Shepard tones to harmonic complex tones, presenting the same interval sizes for each tone type. In both experiments, the ambiguous 6-semitone case for the Shepard tones evoked first-order behavioral variability but high explicit confidence. Pupil dilation was elevated for this condition relative to other conditions in both experiments, suggesting that implicit sensitivity to ambiguity, measured through pupil dilation, may occur without explicit awareness.

Functional ultrasound imaging of ferret auditory cortex reveals a unique neural signature of human speech and music perception

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How have speech and music shaped the human brain? Many signatures of speech and music processing have been observed in non-human animals, raising the question of whether there exist uniquely human mechanisms for processing speech and music. Humans have non-primary neural populations that respond selectively to speech and music compared with both other natural sounds and with synthetic sounds that have matched spectrotemporal modulation statistics ('modulation-matched' sounds), suggesting selectivity for higher-order structure (Norman-Haigneré, 2015/2018). Using functional ultrasound imaging, a cutting-edge high-resolution neuroimaging technique, we tested if similar regions are present in ferrets.

We measured responses from the auditory cortex of passively listening head-fixed ferrets to natural and modulation sounds tested previously in humans. Ferret cortical responses recapitulated many of the response patterns observed in humans. Interestingly, we observed speech selective regions in the ferret auditory cortex. However, and contrary to the real speech- and music-selective response components observed in human non-primary regions, ferret auditory cortex did not show selective responses to natural vs. modulation-matched sounds. These findings suggest that human cortical organization has diverged from other species in non-primary auditory cortex due to the need to represent higher-order structure in speech and music.

Because speech and music are not ecologically relevant sounds for ferrets, we wanted to test whether ferret auditory cortex could discriminate between ferret pup vocalizations and their corresponding model-matched versions. We observed differences in animal motor activity for original vocalizations compared to model-matched stimuli, indicating that the animal is able to perceptually discriminate these two classes of sounds. We are currently investigating the neural correlates of this capability in auditory cortex responses.

Resistance to noise in the auditory system: where does it come from? An electrophysiological study from cochlear nucleus to primary auditory cortex.

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It is known that human subjects can discriminate speech in quite challenging situations such as the presence of important background noise. Many electrophysiological studies, especially performed at the cortical level, have described the way neuronal responses are altered by noisy conditions and have tried to relate response degradation at the neuronal level and behavioral performance. However, very few of these studies have compared the way neuronal responses are impacted at the different levels of the auditory system. Here, we compare the abilities of neurons recorded from primary auditory cortex down to cochlear nucleus to discriminate between spectrally similar communication calls.

Four utterances of a guinea pig whistle were presented in quiet and in two types of frozen noise (a stationary noise and a chorus noise) at 3 SNR levels (+10dB, 0 and -10dB). The vocalizations were presented while recording evoked responses in primary auditory cortex (AI, n=354), the ventral division of auditory thalamus (MGv, n=262), the central nucleus of inferior colliculus (CNIC, n=386) and cochlear nucleus (CN, n=499). Both in stationary and in chorus noise, neuronal responses were degraded but in each structure there was a large diversity of effects from responses already largely attenuated as early as the SNR value of +10dB, up to detectable responses at SNR value of 0dB. In all structures, the stationary noise strongly attenuated the mean evoked firing rate, the trial-to-trial reliability of neuronal responses and the neuronal discriminative abilities indexed by computing Mutual Information (MI). Based on the group data, the best discriminative abilities were observed in the CNIC with a non-significant decrease in MI mean value up to SNR level of 0dB. In the chorus noise, the evoked firing rate, the trial-to-trial reliability of neuronal responses were also decreased in all structures, but the average MI value was only slightly decreased due to the presence of spectro-temporal cues in the frozen chorus noise. Despite this, the subcortical neurons were, on average, better in discriminating the four target whistles than the cortical ones.

These data indicate that the discrimination between spectrally similar vocalizations can be achieved in noisy conditions at all levels of the auditory system but seems more prominent in the CNIC.

Response strength and response spread in primary auditory cortex using different pulse shapes with cochlear implants: Is there a optimal stimulation strategy?

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Many research efforts are still dedicated on improving the coding strategies of cochlear implants (CI). They point out that the stimulation mode, the pulse shape and grounding schemes can exert either moderate, or drastic, consequences on the response strength, spread of excitation and nerve excitability. Several strategies are currently used to implement sound loudness such as increasing the pulse amplitude or the pulse duration. Here, we compared responses from auditory cortex neurons to stimulations delivered through a CI for which several parameters were modified such as the pulse amplitude, the pulse duration and the pulse shape.

Experiments were performed in urethane anesthetized guinea pigs. The tonotopic gradient of the primary auditory cortex (AI) was established by inserting an array of 16 cortical electrodes (2 rows of 8 electrodes separated by 1mm and 350µm within a row). A dedicated stimulation array (300µm) was then inserted in the cochlea (4 electrodes inserted in the 1st basal turn) and its connector was secured on the skull. The cortical electrodes were placed back in auditory cortex at the exact same location as before the CI insertion. Twenty levels of charges were used to activate the auditory nerve through a dedicated stimulation platform. Stimulations were performed by increasing the pulse duration (PD) or pulse amplitude (PA) with either symmetric, asymmetric or ramped pulses. Asymmetric pulse shapes and ramped pulses were also tested for a given level of injected charge. For asymmetrical pulses, the ratio between the pulses phases increased from 1/1 to 1/10; for ramped pulses the angle was from 88 to 57°.

On average, the firing rate (FR) evoked by the pulse duration (PD) and pulse amplitude (PA) strategies was similar and it was often similar with the asymmetric and ramped pulses. However, in many animals FR was lower than with pure tones. The spatial recruitment was often similar with the PA and PD strategy and with asymmetric and ramped pulse, but it was often lower than the one obtained with pure tones played at 75dB SPL.

Group data suggest that equivalent cortical activation can be achieved with PA or PD strategy, with symmetric, asymmetic or ramped pulses. In individual cases, comparing the FR dynamic range and the spatial recruitment obtained with PA, PD, asymmetric pulses or ramped pulses, suggests that a particular strategy or pulse shape is useful for limiting the spatial activation while still eliciting large FR changes.

Probing AM detection in noise with reverse correlation – a pilot study

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Evidence from psycholinguistic studies suggests that the impact of steady noise on envelope perception has two main components: (1) a masking effect: at low SNR, crucial envelope cues in the signal become masked, compromising recognition accuracy; and (2) a confounding effect: intrinsic envelope modulations arising from the filtering of noise into critical bands can be confused with useful modulations in the signal. Reverse correlation techniques are particularly suitable for exploring the confounding effect of noise on perception. However, the only attempt at applying this technique to an AM detection task yielded mixed results [1].

This project aims at estimating psychophysical kernels in the envelope domain for a 4-Hz AM detection task. Here we will present some promising pilot data on 2 participants (5.000 and 2.000 trials), together with simulated data from the Modulation Filterbank (MFB) model. The results reveal that both real and simulated listeners are able to track the modulation peaks in the 4-Hz target. However, unlike the MFB model, human psychophysical kernels are in phase opposition with the ideal template, suggesting that they may represent fluctuations in the target and masker as a short-term signal-to-noise ratio in the envelope power domain (SNR_{env}).

[1] Ardoint, Mamassian & Lorenzi (2007) Internal representation of amplitude modulation revealed by reverse correlation, in proceedings of the 30 th ARO midwinter meeting, Denver, Colorado, USA

Program

MONDAY

12.00	Lunch		
12.55	Panorama	(i)	C. Lorenzi *(Ecole normale supérieure de Paris)*
13.10	Talks	(i) 13.10	E. Gaudrain
		(ii) 13.30	H. Relaño-Iborra
		(iii) 13.50	J. van Schoonhoven
14.10	Panoramas	(i) 14.10	T. Dau *(Technical University of Denmark)*
		(ii) 14.20	M. Akeroyd *(Nottingham University)*
		(iii) 14.30	D. Başkent *(University Medical Center Groningen)*
14.40	Talks	(i) 14.40	L. Simon
		(ii) 15.00	R. Giurda
15.20	Poster session		
16.20	Talks	(i) 16.20	D. Pressnitzer
		(ii) 16.40	H. van der Hoek-Snieders
		(iii) 17.00	D. Verschuren
17.20	Panoramas	(i) 17.20	A. van Wieringen *(Katholieke Universiteit Leuven)*
		(ii) 17.30	N. Dillier *(University Hospital Zürich)*
18.00	Cartesian break		
19.00	Diner @ Concordia		

TUESDAY

9.00	Talks	(i) 09.00	E. Darestani
		(ii) 09.20	J. Märcher-Rørsted
		(iii) 09.40	R. Gransier
		(iv) 10.00	A. Heeringa
10.20	Panoramas	(i) 10.20	B. Kollmeier *(Carl von Ossietzky Universität)*
		(ii) 10.30	W. Dreschler *(Academic Medical Center Amsterdam)* & C. Smits *(VU University Medical Center Amsterdam)*
		(iii) 10.45	J.M. Edeline *(Neuro PSI Paris Saclay)*
10.55	Poster & BOARD		
11.50	Talks	(i) 11.50	F. Denk
		(ii) 12.10	S. Ewert
12.30	Lunch		