ARCHES Meeting 2013

November 18-19, 2013
Paris (France)

Meeting venue

The meeting will take place at the **Espace Pierre Gilles de Gennes**, located in central Paris at the following address:

10 rue Vauquelin
75005 Paris

Dinner

On **Monday 18**, a dinner will take place at 8:00pm the restaurant **Chez Clément Saint Michel,** 9 place St-André-des-Arts, 75005 Paris.

Programme

Monday the 18th of November 2013

12noon-2pm

Short overview talks about the current activities in the individual groups (10 minutes+5mn discussion), with focus on hot topics, new (PhD) projects, new grants, directions, initiatives and potentials for collaboration.

12 - 12:15 Current activities at Ecole normale supérieure, Paris, France
Christian Lorenzi

12:15 - 12:30 Current activities at the Carl von Ossietzky Universität and Hörzentrum Oldenburg (Oldenburg, Germany)
Birger Kollmeier

12:30 - 12:45 Current activities at the Center for Applied Hearing Research, Lingby, Denmark
Torsten Dau

12:45 - 13:00 Current activities at ExpORL, Leuven, Belgium
Jan Wouters

13:00 - 13:15 Current activities at the Institute of Hearing Research, Nottingham, UK
Christian Füllgrabe

13:15 - 13:30 Current activities at the Laboratory of Experimental Audiology, Zürich, Switzerland
Norbert Dillier

13:30 - 13:45 Current activities at the Department of Clinical and Experimental Audiology /Academic Medical Center (Amsterdam, The Netherlands)
Wouter Dreschler

13:45 - 14:00 Current activities at the VU University Medical Center, Department of Otolaryngology, Head and Neck Surgery (Amsterdam, The Netherlands)
Cas Smits

2pm-5.20pm

"Twin contributions" / joined talks from ARCHES groups on selected topics (15mn+5mn questions)

Auditory modelling:

14:00 - 14:20
Model-based characterization and compensation of sensorineural hearing loss
Steffen Kortlang, Birger Kollmeier, Stephan Ewert

14:20 - 14:40 Significance of auditory filtering and compression on the statistical representation of sound textures
Richard McWalter, Torsten Dau

Temporal processes and Ageing effects:

14:40 - 15 Is cortical phase locking of ongoing neural oscillations to speech envelope modulations impaired in dyslexia?
Astrid De Vos, Robert Luke, Hanne Poelmans, Michael Hofmann, Jolijn Vanderauwera, Maaike Vandermosten, Pol Ghesquière, Jan Wouters

15 - 15:20 Neural temporal processing in normal hearing aging persons
Tine Goossens, Charlotte Vercammen, Jan Wouters, Astrid van Wieringen

15:20 - 15:40 Beyond audibility - Contributions of central auditory and cognitive processing to speech perception across the adult lifespan
C. Füllgrabe

15:40 - 16 Coffee break

Binaural processes:

16 - 16:20 Experimental evidence for a cochlear source of the precedence effect
Federica Bianchi, Sarah Verhulst, Torsten Dau

16:20 - 16:40 A statistical approach for the estimation of the direct-to-reverberant-ratio from dual-channel signals
Eleftheria Georganti, John Mourjopoulos, Norbert Dillier

Human electrophysiology:

16:40 - 17 Using Auditory Steady State Responses to characterize CI channel performance
Michael Hofmann, Robert Luke, Lieselot Van Deun, Astrid van Wieringen, Jan Wouters

17 - 17:20 Binaural Evoked Potentials in Young Normal Hearing Adults: Objective Measures of the Binaural System
Charlotte Vercammen, Jan Wouters, Astrid van Wieringen, Tom Francart

Dinner at Chez Clément

Tuesday the 19th of November 2013

8.30am-10am

8:30 - 9:15 Posters

Neural correlates of pitch salience using fMRI
Federica Bianchi, Sébastien Santurette, Torsten Dau, Jens Hjortkjær, Hartwig Siebner

A dynamic compression algorithm for loudness compensation of broad- and narrow-band sounds
Stephan Ewert, Dirk Oetting, Volker Hohmann, Jens-E. Appell

Interaural place-mismatch estimation with two-formant vowels in unilateral cochlear-implant users
Francois Guerit, Sébastien Santurette, Josef Chalupper, Iris Arweiler, Torsten Dau

Analysis of processing speed during sentence comprehension using eye-fixations recorded per Eye-Tracking and Electrooculography (EOG)
Jana Mueller, Dorothea Wendt, Thomas Brand, Birger Kollmeier

Measuring cochlear nonlinearity in the clinic
Michal Fereczkowski, Ewen MacDonald, Torsten Dau

Development of speech perception and spectro-temporal modulation processing
Laurianne Cabrera, Feng-Ming Tsao, Josiane Bertoncini & Christian Lorenzi

Effects of noise reduction on AM discrimination for normal-hearing and hearing-impaired listeners: psychophysical and modeling data
David Timothy Ives, Sridhar Kalluri, Olaf Stelcyk, Stanley Sheft, Christian Lorenzi

Music exposure through mp3-players and the risk of music-induced hearing loss among adolescents, Marya Sheikh Rashid

Improving the Performance of Cochlear Implants by Encoding Temporal Fine Structures and Using Fewer Number of Samples
Sonia Tabibi, Hamed Sadjedi, Norbert Dillier

Age-related slowing in processing time despite perfect speech intelligibility
C. Füllgrabe, M.A. Stone and B.C.J. Moore

Cross-modal influences on perceived lateral position: "attentional repulsion" in the auditory and visual domains
N. Gama, N. Gama, I. Wiggins, A. Palmer and D. Hartley

Speech perception:

9:15 - 9:30 The influence of masker types and cognitive capacity on speech reception and cognitive load (peak pupil dilation) in adults with normal hearing and hearing impairment.
Thomas Koelewijn, Adriana Zekveld, Sophia Kramer

9:30 - 9:45 Is context always helpful? Speech intelligibility measurements with the OLACS
Verena Uslar, Birger Kollmeier, Thomas Brand

Hearing aids:

9:45 - 10 A Self-fit hearing aid
Monique Boymans, Wouter A. Dreschler

10 - 10:15 A profiling system for the selection of hearing aids in the Netherlands
Wouter A. Dreschler

10:15 - 10:30 Coffee break

10:30 am - 11:30 am

Board meeting (1 hour)

Meeting of the PhD students from the different sites which will run in parallel to the board meeting (1 hour)

Lunch at the Espace Pierre Gilles de Gennes

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Model-based characterization and compensation of sensorineural hearing loss

Steffen Kortlang 1,*, @ , Birger Kollmeier 1, @ , Stephan Ewert 1, @

1 : Medizinische Physik, Universität Oldenburg and Cluster of Excellence "Hearing4all", Oldenburg, Germany
* : Corresponding author

A model-based, fast-acting dynamic compression algorithm (MDC2) is introduced which aims at restoring outer hair cell (OHC) functionality by approximating the normal-hearing (NH) basilar membrane input-output function in hearing-impaired listeners. The algorithm was fitted by estimating low-level gain loss (OHC loss) from adaptive categorical loudness scaling data and audiometric thresholds based on Ewert and Grimm [in: proc. ISAAR (2012), 393] and Jürgens et al. [Hear. Res. 270, 177 (2011)]. Aided speech intelligibility was measured in stationary and fluctuating noise and related to conventional compression strategies. Even if audibility and loudness perception can be restored by the proposed and other dynamic compression algorithms, listeners with sensorineural hearing loss generally suffer from a reduced ability to discriminate sounds in complex acoustic scenes. Such supra-threshold processing disorders may not solely be explained by the consequences of OHC damage such as broadening of the auditory filters. Additionally, loss or degeneration of inner hair cells (IHC) including damage of retro-cochlear stages of the auditory pathway may play a role. Such damage would affect the "neural coding fidelity" of the sound waveform resulting in loss of temporal fine-structure information at low frequencies and temporal envelope information at higher frequencies. It has been recently shown that such damage can be present and that temporal coding can be affected even if the audiogram is unremarkable, e.g., in elderly persons. Consequently, differential diagnostics of OHC and IHC damage and understanding of their role in supra-threshold signal processing might help improve hearing aid algorithms and fitting.

Here diagnostics of IHC and OHC damage was investigated, by a series of psychoacoustic measurements including random low-rate frequency modulation (FM) detection thresholds in quiet and background noise. Six young NH listeners, six older NH listeners, and eleven hearing impaired listeners participated. Results confirmed that FM was degraded in the latter groups. Particularly at low frequencies, where audiometric thresholds and filter bandwidths were found to be most similar across groups, differences in FM sensitivity were observed. Two independent mechanisms can be assumed to underly FM detection: i) At small modulation and carrier frequencies, temporal coding of fluctuations in the phase-locked timing of neural spikes plays a role. ii) At higher rates, FM detection is thought to be based primarily on FM-induced amplitude modulation. Both cues were modeled at the level of the auditory nerve (AN) using spike trains generated by a probabilistic auditory model [Meddis, J Acoust Soc Am 119, 406 (2006)]. IHC and OHC damage were incorporated and adapted to predict the psychoacoustic data. By reducing the number of AN fibers, the neural coding fidelity was diminished (introducing "internal noise"), while filter broadening as consequence of OHC damage increased the effect of "external noise" in conditions with noise maskers. The model was able to mimic some effects observed in the data. Future steps are further refinement of the model and assessment of aided performance in the hearing-impared model compared to the normal hearing model.

This work was supported by BMBF 13EZ1127D ("Model-based hearing systems").

Significance of auditory filtering and compression on the statistical representation of sound textures

Richard Mcwalter 1, @ , Torsten Dau 1, @
1 : Technical University of Denmark (DTU)

Background
Often referred to as noise, sound textures are readily identifiable and differentiable and have been shown valuable for investigating peripheral auditory processing [McDermott et al., 2011]. It was found that convincing sound textures could be synthesized based on a small set of statistics measured from a reasonably simple auditory model. The present investigation focused on the variation in the statistical representation of sound textures due to the design of the auditory filterbank and compression.
Method
Synthesis began with the decomposition of a target sound texture with a biologically inspired auditory model. The model includes four main processing stages; (i) an auditory filterbank which captures the frequency selectivity of the peripheral auditory system, (ii) compression derived from psychophysical data, (iii) envelope extraction and finally (iv) a modulation filter-bank to capture envelope fluctuations. A set of statistics were computed at different stages of the auditory model, including marginal statistics and cross-band correlation statistics at the output of the compressed sub-band envelope and variance and cross-band correlation statistics at the output of the modulation filterbank. The original sound texture statistics were then imposed on a Gaussian-noise input to yield a new version of the sound texture with the same perceptual qualities.
Three variants of the auditory filterbank and compression were implemented; (i) an ERB-spaced bandpass filterbank [Glasberg et al. 1990] with a static non-linear compression [Harte et al. 2005], (ii) a gammatone filterbank [Patterson et al., 1987] with frequency dependent static non-linear compression [Lopez-Proveda et al. 2003], and (iii) a dynamic compressive gammachirp auditory filterbank [Irino et al. 2006].
Results
An analysis of the statistical representation of several sound textures showed a difference across the three models. This was observed particularly for the first two marginal moments of the subband envelope, the correlation statistics and the modulation variance. It appeared that the higher marginal moments are not significantly affected by the change in auditory filter shape or compression. The variation in statistics was caused primarily by the change in compression across the three models.
The synthesis model was constructed such that any auditory filter or non-linearity applied during the analysis process is reversed. Therefore, if a perceptually significant feature is captured by the statistics, it would result in a more realistic synthesis of the sound texture. Unfortunately, this improvement has not yet been realized and no perceptual benefit is apparent by using a more advanced auditory model.
Conclusion
Preliminary results suggest that, although there is a difference in the statistical representation, no significant perceptual difference can be observed in the synthetic textures. Further investigation is required to confirm the subjective significance of auditory filtering and compression on the synthesis of sound textures. This approach, however, does offer a framework for the analysis of more temporally complex sounds and may account for the response of the auditory filter shape and compression to significant changes in a time domain signals' amplitude.

Is cortical phase locking of ongoing neural oscillations to speech envelope modulations impaired in dyslexia?

Astrid De Vos 1, @ , Robert Luke 1, @ , Hanne Poelmans 1, @ , Michael Hofmann 1, @ , Jolijn Vanderauwera 2, @ , Maaike Vandermosten 2, @ , Pol Ghesquière 2, @ , Jan Wouters 1, @

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Temporal information in speech, represented as amplitude modulations in the speech envelope, is crucial for accurate speech intelligibility. Although the temporal envelope of speech contains multiple rates of amplitude modulations, especially low-rate modulations seem to be important for speech perception.
A key mechanism in the neural processing of low-rate modulations is the phase locking of ongoing oscillations in the brain to the periodicity of a modulated stimulus. This synchronizing process, in which the phase of the brain oscillations shifts to align with the phase of the periodic stimulus, results in significantly improved neural encoding of the temporal cues relevant for speech perception.

The present study aims to examine the phase locking of ongoing neural oscillations in a broad frequency range including theta, alpha, beta and gamma oscillation rates. The main research question is whether neural phase locking to modulations relevant for speech perception in dyslexic readers differs from normal readers. Given the importance of low-rate modulations for accurate speech intelligibility, differences in the cortical processing of slow temporal variations could relate to the core phonological processing problems found in children and adults with dyslexia.

To investigate neural phase locking, auditory steady-state responses (ASSRs) were recorded in a group of normal-reading and dyslexic adolescents. Continuous speech-weighted noise stimuli were 100% amplitude modulated at modulation frequencies near 4 Hz (theta rhythm), 10 Hz (alpha rhythm), 20 Hz (beta rhythm) and 40 Hz (gamma rhythm). As a reference condition, also 80 Hz modulation was included. Stimuli were presented at 70 dB SPL in three modalities: monaurally to the left ear, monaurally to the right ear and bilaterally to both ears. Responses were recorded using a high-density 64-electrode array mounted in head caps.

Preliminary findings will be discussed at the meeting. The results will contribute to the understanding of neural processing of a broad range of modulations in both normal and dyslexic readers.

Neural temporal processing in normal hearing aging persons

Tine Goossens 1, *, @ , Charlotte Vercammen 1, @ , Jan Wouters 1, @ , Astrid Van Wieringen 1, @

1 : Experimental Otorhinolaryngology (ExpORL)
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* : Corresponding author

Speech understanding difficulties – especially in the presence of background noise - are a common complaint in elderly people and are apparent in middle-aged people as well. The age-related decline in speech understanding is presumably a result of changes in both peripheral hearing sensitivity and central auditory processing, including temporal processing. In the present study, we investigate (changes in) neural temporal processing with increasing age at different stages along the auditory pathway (brainstem to cortex). In order to control for peripheral hearing impairment this is done in carefully selected cohorts of persons who have normal hearing thresholds.

Temporal processing is achieved by oscillatory neural activity that is synchronized to the modulations of auditory signals. In order to investigate neural temporal processing at different stages of the auditory pathway (brainstem to cortex) the modulation frequency of the stimulus is varied. Temporal processing of low-frequent modulations (⩽ 20 Hz) is crucial for speech recognition and is presumably processed more cortically than high-frequent modulations (⩾ 40 Hz).

To investigate the impact of age on neural temporal processing, irrespective of peripheral hearing loss, three normal hearing age cohorts are included in the study: young (20-30 years), middle-aged (50-60 years) and elderly (70-80 years). They all have audiometric thresholds ⩽ 25 dB HL from 125 Hz up to and including 4000 Hz in both ears. They are stimulated with octave bands of white noise centered at 1 kHz at 70 dB SPL. The noise is amplitude modulated by 4, 20, 40 and 80 Hz and presented unilaterally left, unilaterally right and bilaterally. By means of a 64-electrode EEG set-up auditory steady-state responses (ASSRs) are recorded as a measure of the synchronized neural activity.

Data appear to show changes in ASSR response strengths, and thus in neural temporal processing, across the three age cohorts. Because all subjects have normal peripheral hearing thresholds, these changes can be contributed to aging per se. The results will be discussed at the meeting.

Beyond audibility - Contributions of central auditory and cognitive processing to speech perception across the adult lifespan

C. Füllgrabe
MRC Institute of Hearing Research, Nottingham, UK

The increasing life expectancy in most Western countries raises the question of the impact of aging on the individual's quality of life in the future as well as society's cost in providing adequate health care to respond to the specific needs associated with this demographic change. One consequence of aging is reduced speech comprehension in the presence of background noise. This not only constitutes a social handicap for the affected person but may also accelerate cognitive decline, thereby representing a serious public-health issue.

Currently, the diagnosis of such deficits is mainly (and in some countries even exclusively) based on a reduction in the ability to detect sounds as measured by pure-tone audiometry. Accordingly, audiological rehabilitation via the use of hearing aids aims to improve speech intelligibility by restoring, at least partially, normal audibility of speech sounds.

However, several observations indicate that this audibility based approach to speech intelligibility is incomplete. For example, even after providing amplification to compensate for peripheral hearing loss, hearing-impaired listeners rarely show the improvements in intelligibility that would be predicted based on the audibility of the speech signal. In addition, older listeners perform more poorly than young counterparts on speech-in-noise tasks even when peripheral sensitivity (i.e., audibility) is matched. Hence, to remediate age-dependent declines in speech comprehension more successfully, it seems important to acknowledge and quantify the effect of age on central (i.e., retro-cochlear) auditory processing and on cognitive and linguistic abilities involved in speech comprehension.

The aim of this presentation is provide an overview of the author's current line of research by briefly presenting several cross-sectional studies on speech perception across the adult lifespan. A unique cohort of participants was used to investigate age-dependent changes in central auditory and cognitive processing abilities: participants were sampled continuously and (fairly) evenly from the entire adult lifespan (at least 15 participants per age decade between 18 and 91 years) to establish when during adulthood declines first become apparent; all participants had clinically normal hearing (between 125 and, at least, 4000 Hz) to reduce the contribution of peripheral decline to speech-perception deficits. Possible future studies using the same cohort will be outlined.

Acknowledgements
Work supported by the Oticon Foundation and British Society of Audiology.

Age-related slowing in processing time despite perfect speech intelligibility

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It is well established that age-dependent peripheral hearing loss results in a decline in speech intelligibility in older listeners. Less is known about how aging – presumably through changes in "cognitive functions" – affects the speed with which speech is processed. Indeed, it is possible that, even when intelligibility is perfect, older listeners might require more processing time than younger listeners. Slowed processing is likely to impact on speech identification in real-world listening situations where discourse is continuous. However, it is generally not captured by "conventional" performance tasks, assessing untimed identification scores of short speech tokens presented at slow rates.
To study age-related retro-cochlear changes in speech identification, performance in terms of identification scores and reaction times (RTs) was assessed for 28 young (mean age = 22 yrs) and 32 older (mean age = 67 yrs) participants with bilaterally normal hearing sensitivity (i.e., audiometric thresholds ≤20 dB HL between 0.125 and 6 kHz). Participants had to identify sentences from the Corporate Response Measure (CRM) corpus, produced by a male speaker. Responses were made by pressing virtual buttons on a touch screen. The target speech was presented either in quiet or in the presence of different interfering backgrounds at 0-dB signal-to-noise ratio: (i) a "steady" speech-shaped noise, (ii) prose read by another (fe)male speaker, (iii) as in (ii) but time-reversed and (iv) CRM sentences spoken by a female speaker using different keywords than the target talker. All participants also completed the "Map Search" test (a measure of selective attention and speed).
The results show that younger and older listeners achieved nearly identical levels of speech intelligibility in quiet and in the different interfering backgrounds. However, RTs were generally slower in the older group. More importantly, the presence of an interfering background resulted in significantly more slowing in RT for entirely correct sentences (relative to the quiet condition) in the older than the younger listeners but only when the background contained speech. Age and cognitive performance correlated significantly with the amount of RT cost produced by the interfering speech backgrounds; the correlation with cognitive performance remained significant even after partialling out age but only for the CRM-on-CRM condition.

Acknowledgements
Supported by the British Society of Audiology and MRC UK.

Cross-modal influences on perceived lateral position: "attentional repulsion" in the auditory and visual domains

N. Gama*§, I. Wiggins⌧, A. Palmer§ and D. Hartley*

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⌧NIHR Nottingham Hearing Biomedical Research Unit, Nottingham, UK

Focusing attention on a preceding visual cue can cause changes in the perceived lateral position of a sebsquently presented target. This effect was demonstrated almost two decades ago, where participants were asked to judge the lateral displacement of one vertical line relative to another (vernier display paradigm). Suzuki and Cavanagh (1997) described this as "attentional repulsion" and it has been studied extensively in the visual domain. Present research is focused on further exploring this paradigm using cues and targets of different modalities. For example, a similar effect has been reported when the preceding cue was a laterally presented brief sound, instead of a visual cue (Arnott and Goodale, 2006). In the auditory domain, this effect is also present using a Stimulus Timing Dependant Plasticity paradigm (STDP; Dahmen et al, 2010), that follows the Hebbian theory of associative synapse strengthening, ruling out mechanisms of cognitive attention. The ambiguity of whether this effect is due to higher order attentional mechanisms or in fact low level multisensory convergence motivates a healthy debate and research aimed at finding neural and behavioural substrates for these effects. However, past studies of auditory alone and cross modal cues and targets have used a variety of stimulus types and experimental setups, and hence the results are not directly comparable to those reported in the visual domain. Here, we aimed to replicate the "attentional repulsion effect" in the visual domain, to determine whether there is a directly corresponding effect in the auditory domain, and to investigate whether the effect also occurs cross-modally when target and cue are presented in different sensory modalities. In their original report, Suzuki and Cavanagh (1997) stressed that the effect was strongly dependent on the onset asynchrony between cue and target. Thus, we parametrically varied the delay between cue and target in the present study. A repulsion effect is observable, which seems to be present across the six delay conditions. Furthermore, preliminary data seems to suggest an attraction in lateral judgments in one cross-modal condition.

Suzuki, S., & Cavanagh, P. (1997). Focused attention distorts visual space: an attentional repulsion effect. Journal of experimental psychology. Human perception and performance, 23(2), 443–63. Retrieved from http://www.ncbi.nlm.nih.gov/pubmed/9104004
Arnott, S. R., & Goodale, M. a. (2006). Distorting visual space with sound. Vision research, 46(10), 1553–8. doi:10.1016/j.visres.2005.11.020
Dahmen, J. C., Keating, P., Nodal, F. R., Schulz, A. L., & King, A. J. (2010). Adaptation to stimulus statistics in the perception and neural representation of auditory space. Neuron, 66(6), 937–48. doi:10.1016/j.neuron.2010.05.018

Experimental evidence for a cochlear source of the precedence effect

Federica Bianchi [1, @], Sarah Verhulst [2, @], Torsten Dau [1, @]
1 : Technical University of Denmark
2 : Carl von Ossietzky University

The precedence effect (PE) refers to the dominance of directional information carried by a direct sound (lead) over the spatial information contained in its multiple reflections (lags) in sound localization. Although the processes underlying the PE have been largely investigated, the extent to which peripheral versus central auditory processes contribute to this perceptual phenomenon has remained unclear. The present study investigated the contribution of peripheral processing to the PE through a comparison of physiological and psychoacoustical data in the same human listeners. The psychoacoustical experiments, comprising a fusion task, an interaural time difference (ITD) detection task and a lateralization task, demonstrated a time range from 1 to 4.6-5 ms, in which the PE operated (precedence window). Click-evoked otoacoustic emissions (CEOAEs) were recorded in both ears to investigate the lead-lag interactions at the level of the basilar membrane (BM) in the cochlea. The CEOAE-derived peripheral and monaural lag-suppression was largest for ICIs of 1-4 ms. Auditory evoked brainstem responses (ABRs) were used to investigate monaural and binaural lag-suppression at the brainstem level. The responses to monaural stimulation reflected the peripheral lag-suppression observed in the CEOAE results, while the binaural brainstem responses did not show any substantial contribution of binaural processes to monaural lag-suppression. The results demonstrated that the lag-suppression occurring at the BM in a time range from 1 to 4 ms, as indicated by the suppression of the lag-CEOAE, was the source of the reduction in the lag-ABRs and a possible peripheral contributor to the PE for click stimuli.

A statistical approach for the estimation of the direct-to-reverberant-ratio from dual-channel signals

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Reverberation has a significant influence on speech communication, sound localization, and auditory perception in general and speech understanding is degraded in highly reverberant environments [1]. A parameter that is related to the amount of reverberation in a signal is the direct-to-reverberant ratio (DRR). The DRR may be used in the context of hearing aids, as a tuning parameter for signal enhancement methods, such as speech dereverberation. The DRR may be also used for the estimation of the distance between a sound source and a receiver(s), thus potentially assisting auditory scene analysis methods, which are utilized in many modern hearing aids. Intelligent hearing aid processing could potentially attenuate noises, echoes and the sounds of competing talkers, while amplifying a target voice, based on the analysis of the auditory environment.
The DRR can be typically extracted from the room impulse response, but in practice room responses are not always available, since intrusive measurements within the rooms are required. Therefore, considerable interest has been shown in methods that can estimate useful acoustical parameters blindly from reverberant signals [2-3]. Various modern applications (i.e. hearing aids) employ signals obtained from a pair of microphones, usually spaced at a short distance apart (typically between 10 to 20 cm). Thus, in this study such a typical dual-channel signal reception scenario is examined in order to extract the room transfer function DRR from such signals. This study intends to extend previous findings [4-6] regarding the spectral standard deviation of single-channel room transfer function and its relationship to the DRR to the dual-channel scenario. A relationship for the standard deviation as a function of the DRR is derived, obtained from the difference of the magnitude spectra of the omni-pair recorded reverberant signals. Based on the proposed relationship, a model for the estimation of the DRR from dual-channel reverberant signals is proposed, tested, and validated using different types of signals (speech/music), recorded in five different rooms and for various source/receiver distances. Finally, several aspects and restrictions, related to the extension of the proposed model for the generic dual-channel scenario to a binaural scenario (relevant to a hearing aid context) are discussed.

References:

[1] Náběլek A., and Dagenais P. (1986). "Vowel errors in noise and in reverberation by hearing-impaired listeners," J. Acoust. Soc. Am. 80, 741–748.
[2] Y. Hioka, K. Niwa, S. Sakauchi, K. Furuya, and Y. haneda, "Estimating direct-to-reverberant energy ratio using D/R spatial correlation matrix model", IEEE Trans. On Audio, Speech and Language Process., 19(8), 2374-2384 (2011).
[3] M. Kuster, "Estimating the direct-to-reverberant energy ratio from the coherence between coincident pressure and particle velocity", J. Acoust. Soc. Am. 130, 3781 (2011).
[4] J. J. Jetzt, "Critical distance measurement of rooms from the sound energy spectral responses", J. Acoust. Soc. Am. , Vol. 65, 1204-1211 (1979).
[5] M. Schroeder, ``The statistics of frequency responses in large rooms (in German)'', Acustica, Vol. 4, 594-600 (1954).
[6] K. J. Ebeling, ``Influence of direct sound on the fluctuations of the room spectral response'', J. Acoust. Soc. Am., Vol. 68, 1206-1207 (1980).

Using Auditory Steady State Responses to characterize CI channel performance

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Cochlear implants (CIs) are mainly programmed by defining threshold and comfort levels for each channel based on the perceived loudness for the patient. More advanced fitting paradigms based on performance would be desirable but are not feasible within the time constraints of clinical practice. In this study, we aim to objectively characterize stimulation channels to allow for the development of faster and more accurate fitting procedures.

EASSRs to amplitude modulated pulse trains have been recorded in 5 users of Cochlear Nucleus CIs using a BioSemi 64 channel scalp electrode EEG 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 EASSR amplitude, phase and signal to noise ratio were investigated.

EASSR responses were detected and show variation across CI stimulation sites. Work is ongoing to relate response differences with channel parameters and behavioral measures of modulation detection and speech understanding.

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

Binaural Evoked Potentials in Young Normal Hearing Adults: Objective Measures of the Binaural System

Charlotte Vercammen 1, @, Jan Wouters 1, @ ,
Astrid Van Wieringen 1, @ ,Tom Francart 1, @

1 : Experimental Otorhinolaryngology (ExpORL)
O&N II Herestraat 49 box 721 3000 Leuven - Belgium

Normal hearing participants integrate input from two ears. This allows them to understand speech (in noise) or attend to one listener in the presence of others, even under acoustically challenging conditions. The binaural system accounts for sound localization as well, by processing and integrating differences in phase (low frequencies) or intensity (high frequencies) between sounds arriving at the left and right ear.

The goal of the present study is to determine objective measures for binaural processing, thereby replicating and extending work by Ross et al. [(2008) A Novel Type of Auditory Responses: Temporal Dynamics of 40-Hz Steady-State Responses Induced by Changes in Sound Localization. Journal of Neurophysiology, 100, 1265-1277)]. Neural temporal coding is investigated by means of multiple-electrode Auditory Steady State Responses (ASSRs) and change responses detected in the EEG time domain and spectrum. Six young normal hearing participants listened passively to sinusoidal amplitude-modulated tones. Individual stimuli were 4 seconds in duration, with a phase shift in the sound carrier of -90° in the left and +90° in the right ear – resulting in an overall phase shift of 180° – after two seconds (slow protocol) or recurring every 400 milliseconds (fast protocol). ASSRs to different modulation frequencies (20, 40 and 80 Hz) were used to tap into different neural generators and phase responses to the IPD change were expected to elicit a short distortion in the synchronization of the ASSRs. Carrier frequencies ranging from 500 up to 1500 Hz were used, showing an upper level for phase response detection.

The electrophysiological results of the 40 Hz modulation frequency were similar to the ones reported by Ross et al. [(2008) A Novel Type of Auditory Responses: Temporal Dynamics of 40-Hz Steady-State Responses Induced by Changes in Sound Localization. Journal of Neurophysiology, 100, 1265-1277)]. Preliminary data also shows similar findings for a 20 Hz modulation frequency, which is believed to tap into more cortical sources than the 40 Hz modulation frequency. This is particularly interesting as it is believed to represent phonemic rate in speech. However, a 80 Hz modulation frequency (brainstem) fails to show a clear response so far. More data will be presented at the meeting.

The influence of masker types and cognitive capacity on speech reception and cognitive load (peak pupil dilation) in adults with normal hearing and hearing impairment.

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A major complaint of people with hearing loss and sometimes of individuals with normal PTAs is the increase in effort while listening to speech in noise. Previous research indicates a relation between cognitive capacity and the speech reception threshold (SRT). People benefit from larger working memory capacity (WMC) when it comes to speech perception especially in more complex listening situations. The current study investigated how different kinds of background noise and WMC affect cognitive load, while listening to speech, in adults with normal hearing and hearing impairment
In this study the pupil response, a measure of cognitive load, was recorded in thirty-two normal hearing (mean age 51 years) and thirty-two hearing-impaired participants (mean age 59 years) while listening to sentences masked by fluctuating noise or a single-talker at either 50% or 84% intelligibility. Additionally, participants performed tests measuring WMC, inhibition of interfering information in a working memory task, and the ability to read partially masked text.
Normal hearing participants, showed slightly better SRTs for speech masked by a single-talker compared to fluctuating noise. The hearing-impaired group displayed worse SRTs for speech masked by a single-talker compared to fluctuating noise. Interestingly, for both groups the peak pupil dilations were larger when listening to speech masked by a single-talker as compared to fluctuating noise when keeping intelligibility level fixed. Regression analyses revealed that larger WMC and better inhibition of irrelevant information related to better SRTs in both participant groups. However, cognitive capacity of hearing-impaired individuals did not explain variance in the pupil response in adverse listening conditions, whereas it did for normally hearing listeners. For normal hearing participants larger WMC, better working memory related inhibition, and a better ability to read partially masked text related to larger peak pupil dilations.
In conclusion, for people with hearing-impairment, like in normal hearing individuals, ignoring interfering speech results in more cognitive load than filtering out meaningless sounds. However, people with normal hearing do not show a drop in performance as a consequence of interfering speech, they even performed better. WMC and the ability to filter out irrelevant information do relate to the SRTs for the single-talker masker conditions. However, these cognitive abilities do not explain interindividual differences in the pupil responses of hearing-impaired individuals. Possible explanations will be discussed.

Is context always helpful? Speech intelligibility measurements with the OLACS

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Typically words or sentence fragments are easier to understand when presented within a sentence context (Miller et al., 1951; Boothroyd & Nittrouer, 1988). This study investigated the influence of linguistic complexity on this phenomenon by using three different sentence types with different levels of linguistic complexity from the Oldenburg Linguistically And Audiologically Controlled Sentences (OLACS, Uslar et al., 2013). The hypothesis was, that embedding sentence fragments into an easy sentence structure increases intelligibility as expected, whereas embedding the same fragment into a more complex or unusual sentence structure decreases intelligibility, thus showing that sentence context might not always be helpful. Discrimination functions for sentence fragments of 40 sentences per sentence type were calculated, both for the fragments presented alone and in a sentence context. Results indicate that intelligibility of the fragments depends on the sentence structure they were embedded in, thus proving that unusual sentence structures can indeed negate the positive effect of sentence context. How these results relate to speech intelligibility models and how this knowledge might help to improve model predictions has to be investigated in the future.

A Self-fit hearing aid.

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"Self-Fit" defines a technology for individual, "user-controlled", adjustment of hearing aids. The method patented describes a system based on user preference responses to pre-processed acoustical stimuli that are collected to automatically calculate best individual amplification parameters.

The "Self-Fit" system results in two different product types:· First-Fit: a subjectively governed search of amplification parameters without the need of audiogram-information. We use a combination of in-situ tone audiometry and in-situ ACALOS. The target group for this application consists of subjects with mild to moderate hearing losses in emerging markets with significant lacks in audiologic infrastructure.

Final-Fit: a "fine tuning tool" (or home-fitting tool) in fitting software to be used as addition to audiogram-based first fit. Fine tuning of audiogram-based "first fit" parameters operates with paired comparisons between pre-processed acoustic stimuli offered for simultaneous comparison through the hearing device. This application delivers a more systematic fine-tuning of hearing devices for the benefit of the hearing impaired

We present some preliminary results of a clinical study in progress, designed to analyse the following aspects of the Self-fit approach:
- Applicability of threshold measurements through in-situ audiometry
- Applicability of in-situ loudness scaling
- Quality of the First-fit
- Feasibility of fine tuning through Self-Fit to Final-Fit

A profiling system for the selection of hearing aids in the Netherlands

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In the Netherlands the reimbursement of hearing aids by Health Insurance Companies is in debate and will be continued only if the selection process is made more transparent. A system is needed that provides an adequate solution: a simple hearing aid when possible and a more complex aid when necessary. In our opinion, scientific literature is not strong enough for an evidence-based approach to be the basis under such a system. An additional complication is that the hearing aid characteristics are only partly public domain, because some details are kept secret or are hidden by commercially oriented names and theories.
For this purpose a first model was designed to match a user profile (Human Related Intended Use) to a hearing aid profile (Product Related Intended Use).
The Human Related Intended Use (HRIU) is based on a self-report inventory of the users' disabilities, weighted by environmental factors, auditory tasks, and individual targets for rehabilitation. Objective hearing tests (tone and speech audiograms) are included in the system, but play a minor role. This approach yields an individual HRIU profile with scores on 6 dimensions: detection, speech in quiet, speech in noise, localization, focus, and noise tolerance.
A complete inventory of the hearing aid characteristics of more than 1400 commercially available hearing aids was made. For each of the six dimension described above, the features and characteristics that were assumed to be relevant for that dimension were selected by a panel of independent experts. The scores of a specific hearing aid on relevant items were higher if that hearing aid provided features that were judged to be potentially helpful for the compensation of that particular dimension. This resulted in a 6-dimensional Product Related Intended Use profile (PRIU), representing objective criteria for the complexity for the technology available for each of the six dimensions.
The HRIU-profile determines the degree of complexity and/or sophistication of the hearing aid needed and the PRIU profile is helpful in finding appropriate candidates within the (usually large) selection of hearing aids available.
Post-fitting, a well-standardized evaluation procedure is required, including the same inventory of disabilities. The results show the improvements in the 6 dimensions of auditory functioning. This determines whether the hearing aid is adequate and can be sold. But also it provides well-standardized data to evaluate the basic assumptions and to improve the system based on practice-based evidence.
This approach can only work for very high numbers of fittings. A nation-wide implementation will result in an inclusion of >10.000 cases each month. Valuable data will become available and practice-based evidence will compensate for the problem that an evidence-based approach was not possible in the beginning. The system of hearing aid fitting will become more transparent, the effectiveness of specific features can be analyzed and the individual improvements can be related to expectations based on large-sample populations.

Neural correlates of pitch salience using fMRI

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Neuroimaging studies have investigated the existence of a pitch center in the human brain. While some early studies using iterated-ripple noise (Griffiths et al., 1998) and complex tones (Penagos et al., 2004) reported a consistent activation to pitch in lateral Heschl's gyrus (HG), more recent studies (Hall and Plack, 2009; Barker et al., 2011) did not find significant activation in HG in response to different pitch-evoking stimuli. A general pitch center should not only respond to all pitch-evoking stimuli (pitch constancy), but also covary with pitch salience. Covariation of neural activity with pitch salience was investigated by Penagos et al. (2004), who found that resolved complex tones (strong pitch) elicited higher neural activation than unresolved complex tones (weak pitch) in normal-hearing subjects (NH). However, further investigations did not find such neural correlates of pitch salience (Hall and Plack, 2009; Barker et al., 2011).
In the present study, two imaging paradigms were designed and tested to estimate functional-magnetic-resonance-imaging (fMRI) activation in response to complex tones with varying salience. The pitch salience of complex tones is known to increase as a function of fundamental frequency (F0): small F0s(unresolved complex tones) give rise to a weak pitch, whereas large F0s (resolved complex tones) elicit a salient pitch. The hypothesis is that, if fMRI techniques are able to detect changes in pitch salience, the results should show a difference in activation in response to resolved vs unresolved complex tones.
The first imaging paradigm was similar to that used by Penagos et al. (2004). Resolved and unresolved complex tones with F0s of 100, 200 and 500 Hz, filtered into low- and high-frequency regions, were presented according to a block design consisting of 24 s of stimulation and 24 s of scanning. A repetition-time (TR) of 8 s minimized the interference of scanning noise with stimulation. Preliminary data from two NH listeners suggested that this paradigm might not be sensitive enough to observe differences in individual subjects. In order to increase statistical power, a novel event-related imaging paradigm was thus designed. Complex tones with parametrically-varying F0 were presented in short silent gaps between long scanning blocks. This paradigm might allow to investigate individual differences across NH and HI subjects, as scanning occurs over a longer period and F0 is parametrically varied, while a pitch-discrimination task keeps the subject alert.
Variations in cortical activity measured with this second paradigm will be compared to behavioral estimates of pitch salience (difference limens for F0, F0DLs) in the same listeners. As the difference in F0DLs between resolved and unresolved complex tones is typically large for NH subjects, but may be smaller or negligible in individual HI listeners (Bernstein and Oxenham, 2006), the relationship between objective and behavioral data may help clarify whether the observed cortical activation is correlated to pitch salience, and how it is affected by hearing impairment.

A dynamic compression algorithm for loudness compensation of broad- and narrow-band sounds

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Loudness is a fundamental percept that can be easily assigned to any stimulus. Hearing impaired listeners usually suffer from distorted loudness perception with overly step loudness growth (loudness recruitment) above (elevated) threshold. Adequate loudness perception is one of the key factors related to overall satisfaction with hearing aids. However, about 20% of the hearing-aid users still mention to be dissatisfied particularly with loud sounds. A fundamental problem when attempting to restore loudness perception in hearing impaired listeners are differences in the loudness perception of narrow- and broadband sounds when compared to normal-hearing listeners.

Here, a multi-band dynamic compression algorithm is presented that considers different target gains per band required for narrow- and broad-band signals. The goal is to restore loudness perception independent of signal bandwidth, accounting for absent or residual spectral loudness summation. The algorithm applies the desired gain according to the signal-to-masking ratio (SMR) which measures the masking effect of other bands on the current band. The SMR can be used the estimate the bandwidth of the signal. Loudness perception was assessed by categorical loudness scaling for different signal bandwidths. The proposed algorithm was compared to classical gain prescription rules for a variety of narrow- and broad-band signals (noise and speech) and hearing losses, using a recent loudness model [Chen et al., Hear. Res. 282, 69 (2011)]. The evaluation of this SMR approach with the model showed that the loudness perception of hearing-impaired listeners can be restored to the loudness perception of a normal-hearing listener for signals with different bandwidths. However, inconsistencies between the individual measured loudness function using the categorical loudness scaling procedure and the model predictions were found. The available model parameters, being i) hearing threshold level, ii) outer, and iii) inner hair-cell loss, were not sufficient to fit the model to the individual narrow-band loudness perceptions.

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Interaural place-mismatch estimation with two-formant vowels in unilateral cochlear-implant users

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Background: For patients with one cochlear implant (CI) and residual hearing in the opposite ear, a default frequency-to-electrode map is typically used despite large individual differences in electrode-array insertion depth. This non-individualized fitting rationale might partly explain the variability in long-term speech-reception benefit among CI users. Knowledge about the electrode-array location is thus crucial for adequate fitting. Although electrode location can theoretically be determined from CT scans, these are often unavailable in audiological practice. Moreover, existing behavioral procedures such as interaural pitch-matching are rather tedious and time-consuming. Here, an alternative method using two-formant vowels was developed and tested.

Methods: Eight normal-hearing (NH) listeners were presented synthesized two-formant vowels embedded between consonants /t/ and /k/, with first-formant frequencies (F1) at 250 and 400 Hz and second-formant frequencies (F2) between 600 and 2200 Hz. F1 was presented unaltered to the left ear, while F2 was presented to the right ear via a vocoder system simulating 3 different CI insertion depths. In each condition, the listeners indicated in a forced-choice task which of 6 vowels they perceived for different [F1, F2] combinations. Ten CI users (5 bimodal and 5 single-sided deaf) performed the same task for F1 presented acoustically to the non-CI ear and F2 presented either acoustically to the same ear or electrically to the CI ear.

Results: After some training, all NH listeners were able to fuse the unaltered F1 and vocoded F2 into a single vowel percept, and vowel distributions could be reliably derived in 7 listeners. Vocoder simulations of reduced CI insertion depth led to clear vowel-distribution shifts in these listeners. However, these shifts were overall smaller than their theoretical value, with high across-subject variability. Vowel distributions could be derived for all CI users in the monaural acoustic condition, indicating an ability to perform the task reliably. Despite this, large individual differences were observed for dichotic bimodal stimulation, with listeners showing either basal or apical shifts, or generally-poor vowel discrimination.

Conclusions: The two-formant-vowel method is a fast and clinic-friendly candidate to derive interaural place mismatches from a simple vowel-recognition task. However, it remains unclear whether the measured “vowel spaces” in CI users are directly related to insertion depth, and whether they are influenced by the ability to fuse acoustic and electric stimuli or habituation to the CI. The comparison of the present results to CT-scan and speech-intelligibility data in the same listeners will shed light on the validity of the proposed method.

Analysis of processing speed during sentence comprehension using eye-fixations recorded per Eye-Tracking and Electrooculography (EOG)

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Wendt and colleagues developed a method for analyzing eye-fixations in order to estimate the speed of sentence processing as a measure of cognitive processing effort during sentence comprehension (Wendt et al. 2012). Their eye-tracking paradigm consists of an acoustic stimulus (spoken sentence) and a visual stimulus (two alternative pictures showing the entities of the spoken sentence with interchanged roles). Both stimuli, i.e. the spoken sentence and the pictures, were presented simultaneously and the participants' task was to identify the target picture (the corresponding picture) by pressing a button as soon as possible after the spoken sentence. During the paradigm participants' eye-fixations were recorded using and eye-tracking device. From the recorded eye-fixations the single target detection amplitude (sTDA, which describes the tendency to fixate towards the target picture as a function of time) and the corresponding disambiguation to decision delay (DDD, which provides an effective measure for processing speed) were calculated. By analyzing the processing speed for sentences that differ in their level of linguistic complexity, a reduced processing speed for more complex sentence structures was observed. To systematical change the level of linguistic complexity, the OLACS (Oldenburger Linguistically and Audiologically Controlled Sentences) corpus was used that provides sentence structures that differ in their linguistic complexity (Uslar et al. 2013). The current study investigated if the eye-tracking paradigm and the corresponding data analysis, developed by Wendt et al., can also be applied using an additional method for the registration of eye-fixations, namely the electrooculography (EOG). EOG provides a recording method, which registered eye movements using two electrodes to measure potential differences between both eyes. The evaluation of the EOG as an alternative method was realized by recording eye-movements with the eye-tracking camera and simultaneously with an electrooculography (EOG) system. Twelve hearing- impaired listeners and five normal-hearing listeners, which were matched in age, participated in the study. From the recorded eye-tracking data and EOG data the sTDA and the corresponding DDD were calculated for each participant. Cross correlations between both obtained sTDAs were about 0.9 across all participants. The results suggest that the EOG method represents a useful and alternative tool to measure the processing speed during sentence comprehension under usage of the method developed by Wendt and colleagues.

References:

Wendt, D., Brand, T., Kollmeier, B. (2012). The influence of linguistic complexity, noise on speech comprehension: Evidence from eye movements. Speech in Noise Workshop, Cardiff, UK.

Verena N. Uslar, Rebecca Carroll, Mirko Hanke, Cornelia Hamann, Esther Ruigendijk, Thomas Brand, and Birger Kollmeier (2013). Development and evaluation of a linguistically and audiologically controlled sentence intelligibility test. J. Acoust. Soc. Am., 134, 4, 3039-3056. DOI: 10.1121/1.4818760.

Measuring cochlear nonlinearity in the clinic

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Researchers, hearing industry representatives and, importantly, clinicians agree that the tools and techniques that are currently used in the clinics are not sufficient to fully characterize hearing deficits. This, in turn, results in suboptimal hearing aid fitting and decreases the overall satisfaction of hearing aid users. Thus, the ability to estimate other characteristics, such as compression and outer hair cell (OHC) related hearing loss, may improve clinical outcomes. Using psychophysical measurements based on forward masking, the compression characteristics of the basilar membrane input-output functions (BM I/O) can be estimated. However, traditional methods using temporal masking (TM) and/or growth of masking (GOM) paradigms take too long to be clinically feasible. Here, we present a new, time-efficient method derived from standard TM methods. By sampling the BM I/O curve more efficiently (i.e., by testing the minimum number of points needed to reliably estimate BM I/O parameters) the new method can estimate compression and OHC-related hearing loss in approximately 5-10 minutes per frequency. Results from four normal hearing and four hearing impaired listeners are compared using both the new and traditional methods. The maximum compression ratios measured with the new method are more consistent among NH listeners and hence more reliable when compared to the results of the standard method.

Development of speech perception and spectro-temporal modulation processing

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The present studies aimed to assess the contribution of basic auditory mechanisms to speech perception during infancy. Speech perception abilities have been explored extensively in young infants over the last decades (see Kuhl, 2004 for a review), and it is now clear that during the first year of life, speech perception undergoes strong modifications influenced by the environmental language. Surprisingly, the exploration of the mechanisms specialized for speech perception and their development has relegated auditory mechanisms to a position of secondary importance. Nonetheless, the ability to perceive speech sounds and to acquire phonological knowledge depends on the efficiency of the auditory system and its development. Recently, psychoacoustic studies conducted with adult listeners have offered a novel description of the auditory mechanisms crucial for speech processing (see Shamma & Lorenzi, 2013). This description emphasizes the role of the gross and fine spectro-temporal modulation (amplitude-modulation (AM) and frequency-modulation (FM)) cues present in the speech signal.

The present set of studies focus on normal-hearing 6- and 10-month-old infants learning different languages (here French or Mandarin). These studies aimed to explore the contribution of basic auditory mechanisms that process the spectro-temporal modulation cues during the second half of the first year.

Infants' ability to discriminate phonetic (voicing and place of articulation) and lexical-tone contrasts was evaluated using a behavioral method (visual habituation) in several conditions. Signal-processing algorithms called "vocoders" have been used to selectively alter the AM and FM content of natural speech signals. Moreover, some experimental conditions in which FM and spectral resolution were reduced have been designed to simulate speech processing by current cochlear-implant (CI) processors.

Altogether, the results showed that: 1) fine spectro-temporal modulation cues (the FM cues and fine spectral details) are not required for the discrimination of voicing and place of articulation for French-learning 6-month-old infants at least in silence; 2) fine spectro-temporal modulation cues are required for lexical-tone discrimination for French- and Mandarin-learning 6-month-old infants; 3) linguistic experience influences the use of the modulation cues related to F0 (or voice-pitch variations) for both young adults and 10-month-old infants learning either French or Mandarin. Specifically, lexical-tone users were more impaired by the degradation of fine spectro-temporal modulation cues than non-users. Thus, the perception of the spectro-temporal modulation cues of speech is plastic (i.e., it is influenced by the listener's perceptual experience). Moreover, simulations of speech processing by CI processors in normal-hearing infants show to what extent spectral and temporal modulation cues are required for the normal development of speech-perception mechanisms.

Effects of noise reduction on AM discrimination for hearing-impaired listeners

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Background:
Noise-reduction (NR) algorithms are employed in digital hearing-aid devices to improve the listening experience of the user in a noisy background. Although these algorithms may increase the signal-to-noise ratio (SNR) in an ideal case, they generally fail to improve speech intelligibility. However, due to the complex nature of speech, it is difficult to disentangle the numerous effects of noise reduction which may underlie the lack of speech benefits.
Ives et al (2012) examined the effect of a NR algorithm on the ability of normally-hearing (NH) listeners to discriminate a basic acoustic feature known to be crucial for speech identification, namely amplitude modulation (AM). They found that NR slightly improved discrimination at the higher SNRs. The goal of the present study was to assess whether the benefit of NR on AM discrimination was present for hearing-impaired (HI) listeners.

Methods:
The discrimination of complex AM patterns was measured for 10 HI listeners and 10 NH listeners using a same-different discrimination task. The stimuli were generated by modulating a pure-tone carrier by a two-component AM modulator with modulation rates centered around 3 Hz. The carrier tone was either 500 Hz or 2 kHz and was fixed within a block. Discrimination was measured for both groups of listeners (NH and HI) at 500 Hz and 2 kHz in the presence of a band-pass filtered Gaussian white noise at an SNR of 12dB. Stimuli were left as such or processed via a NR algorithm based on the spectral subtraction method. The HI listeners had normal hearing (⩽20dB HL) at 500 Hz and a moderate-severe hearing loss (⩾40dB HL) at 2 kHz.

Results:
NR was found to: (i) improve AM discrimination at both 500 Hz and 2 kHz for the NH listeners; (ii) improve AM discrimination at 500 Hz for HI listeners; (iii) have no effect on AM discrimination at 2 kHz for HI listeners.
The stimuli were passed through a basic computational model of the peripheral auditory system. The simulation results suggest that the lack of benefit of NR for the HI listeners at 2 kHz may arise from the combined effects of higher absolute thresholds and reduced cochlear compression. Auditory filter width did not affect performance.

Conclusions:
HI listeners do not benefit from NR for AM discrimination. The results suggest that this lack of benefit may arise from a poor matching between the compression stage and NR algorithm in hearing aids.

Music exposure through mp3-players and the risk of music-induced hearing loss among adolescents

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Introduction
The widespread popularity of listening to music through MP3-players among adolescents has become a major public health concern. Frequently listening to loud music for long periods may result into hearing damage among youngsters. The aim of this research was to provide insight into music-induced hearing loss (MIHL) involving the use of MP3-players among adolescents in the Netherlands.

Research questions
We tried to accurately estimate the number of adolescents (12-24 years) who listen to music through MP3-players in excess of safe music exposure levels. We also tried to explore socio-demographic factors such as age, gender, education level, and music style that potentially predispose adolescents to high music exposure levels.

Methods
A technical model to estimate sound output produced by MP3-players (based on type of MP3-player and earbuds, volume settings, and music style) was developed. This was followed by the development of a risk model for the prediction of music-induced hearing loss after 10, 15, or 20 years of music exposure, derived from ISO-1999. These models were subsequently integrated in the MP3-Check, an online test developed to give insight into music listening habits, including listening duration and volume settings (www.mp3check.nl). Cross-sectional data derived from test results of participants of the MP3-Check were collected for six years, from which individual weekly equivalent music exposure levels were calculated.

Results
An elevated hearing loss risk for 8 to 10% of the adolescents participating in the MP3-Check was predicted, of which about 5% are at high risk of developing MIHL. For teenagers (12-17 years), this predicted risk was higher (8.8%) than for young adults (18-24 years) (5.9%). Older teenagers, boys, lower secondary and vocational education level students, and listeners preferring music styles such as dance and hip hop, are riskier in their music-listening behavior as compared to their counterparts.

Conclusions
The relatively accurate estimation of individual weekly equivalent music exposure levels in combination with a developed risk model for music exposure, suggested that a substantial percentage of adolescents are listening to music through MP3-players in a risky manner. This research provided a further step in understanding the problem of MIHL with a view to the development of targeted health education and MIHL-preventive measures.

Improving the Performance of Cochlear Implants by Encoding Temporal Fine Structures and Using Fewer Number of Samples

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The Cochlear Implant is widely used as a treatment for patients with severe to profound deafness. Direct electrical stimulation of the auditory nerve has been beneficial in restoring useful hearing in cases where acoustic stimulation cannot provide sufficient information for adequate speech perception [1]. Speech processor algorithms are crucial requirements for effective extraction and delivery of speech information to the auditory system [2].
Signal processing strategies purely based on amplitude modulations cause difficulties for cochlear implant users' performance in noise [3]. As a result, other signal processing methods are explored which focus on possible options of encoding spectral and temporal fine structure cues in cochlear implants. One way to encode the temporal fine structure is to extract frequency modulation and then use it to frequency modulate the carrier rate [4]. Experimental evidence has indicated that the normal auditory nerve does not produce a spike train at a fixed rate [3]. Thus, it seems that contemporary signal processing strategies such as Continuous Interleaved Sampling (CIS) which are based on amplitude modulated carrier signals are not optimal solutions.
In recent years strategies have been proposed that can extract frequency modulation in addition to amplitude modulation; Frequency Amplitude Modulation Encoding (FAME) is one of these strategies which was proposed by Nie et al. in 2005. In this method amplitude and frequency modulations are extracted in two parallel pathways for each band. The amplitude modulation pathway extracts the slowly varying envelope, while the frequency modulation pathway extracts the slowly varying frequency modulation [3].
The results of the present study based on spectrogram analyses and questionnaires demonstrate that the FAME strategy contains potentially more acoustic information compared to the CIS strategy. However, it also leads to a higher computational load. Therefore, there is a need for sample reduction techniques which can transmit the information adequately. Three different sample reduction methods are proposed in this study. One method which is proportional to the input signal is considered to yield better results due to the use of fewer samples to synthesize the signal. Eventually these methods may also be useful to determine the optimal stimulus rate for intracochlear electrodes.

References:

[1] K. Arora, R. Dowell and P. Dawson (2012), “Cochlear Implant Stimulation Rates and Speech Perception”, chapter 10, 215-254.
[2] F. Chen and Y.-T. Zhang (2008), “A novel temporal fine structure-based speech synthesis model for cochlear implant”, Elsevier Signal Processing 88, 2693-2699.
[3] K. Nie, G. Stickney and F.-G. Zeng (2005), “Encoding Frequency Modulation to Improve Cochlear Implant Performance in Noise”, IEEE Trans. Biomed. Eng., vol. 52, no. 1, 64-73.
[4] F.-G. Zeng, S. Rebscher, W. Harrison, X. Sun and H. Feng (2008), “Cochlear Implants: System Design, Integration, and Evaluation”, IEEE reviews in Biomed. Eng., vol. 1, 115-142.