6$^{\text{th}}$ ARCHES meeting

Copenhagen — November 26-27, 2012 *www.arches2012.dk*

Dear friends and colleagues, we are pleased to welcome you to the sixth edition of our annual ARCHES meeting in Copenhagen. We are very much looking forward to another inspiring scientific exchange and fruitful auditory discussions between the members of the ARCHES network. We are happy that all ARCHES groups have responded to our invitation and will be represented at this year's meeting. ARCHES has so far been a successful platform to network, informally discuss research topics, and create new collaborations between our labs, and we wish to keep up this dynamic spirit and continue to share ideas in a friendly atmosphere.

In this document you will find practical information as well as the programme and abstracts for the meeting. This year's programme consists of 13 talks and 10 posters, and we would like to thank you for these interesting contributions covering a diverse range of topics. We have grouped oral presentations into 4 different sessions around the themes of perceptual effects of hearing-device signal processing, speech production and recognition in adverse conditions, auditory signal processing and virtual environments, and music-induced hearing loss. We hope you'll enjoy a productive meeting and wish you a pleasant stay in Denmark's capital city!

The ARCHES 2012 organizing committee,
Sébastien Santurette, Caroline van Oosterhout, Torsten Dau

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Practical information

Meeting location

The meeting venue is the Kosmopol Auditorium, located in central Copenhagen at the following address:

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Fiolstræde 44
DK-1171 København K
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The talks and poster presentations will take place in the auditorium (Auditoriet) on the first floor. From the street entrance, ring the Kosmopol bell and walk up the first set of stairs.

Getting there

The meeting venue is conveniently located close to Nørreport station, the largest public transport hub in Copenhagen. Follow these directions:

From Copenhagen Airport (CPH):
Take Metro M2 for a 15-min ride to Nørreport station. From there, it is only a 2-min. walk to Kosmopol. See the map below.

From the Central Station (København H):
Take any northbound S-train from platforms 9-10 (lines A, B, C, E, or H) and get off at the second stop, Nørreport station. From there, it is only a 2-min. walk to Kosmopol. See the map below.

Single fare: DKK 36 (3 zones) from the Airport, DKK 24 (2 zones) from the Central Station. You can use the same ticket for all means of transportation within the ticket's validity period. Sharing a 10-trip discount "clip card" (DKK 145 for 2 zones, DKK 190 for 3 zones) might be a good option if you travel in a group or intend to stay longer. Alternatively, you may want to invest in a City Pass, valid in all central zones including the airport for 24 hours (DKK 75) or 72 hours (DKK 190). For more fare and schedule info, visit `www.journeyplanner.dk` or `www.movia.dk`.

Accommodation

We have arranged a special rate for all ARCHES attendees at Hotel Alexandra, located at the following address:

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H.C. Andersens Boulevard 8
DK-1553 København V
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Room for 1 person: DKK 935 (ca. EUR 126)

Room for 2 persons: DKK 1035 (ca. EUR 139)

The above rates are valid between November 23 and 29, 2012. They include complimentary breakfast and welcome drink, as well as Wi-Fi internet access. Room reservations at these rates can be made online at `www.hotelalexandra.dk` by using promo code "arches2012". Rooms may be canceled free of charge up to 6 pm on the day before arrival.

It is an approx. 12-min. walk from Hotel Alexandra to the meeting venue.

Abstract submission

You may submit research contributions for the 2012 ARCHES meeting online at `www.arches2012.dk`. Talks and poster presentations are the two possible formats. Joint talks between several research groups are highly encouraged. Please limit the length of your abstract to 400 words.

Presentation guidelines

Please limit the duration of your talk to 20 minutes. There will be a 10-minute discussion after each talk. Poster format should be vertical with a maximum width of 100 cm and a maximum height of 150 cm. All posters will be displayed throughout the meeting with 90 minutes of presentation time on both days.

Registration

We kindly ask ARCHES attendees to register online at `www.arches2012.dk`. The registration fee of DKK 745 (EUR 100) includes participation in the meeting and the social event, as well as all meals (lunch and dinner on Monday 26/11, lunch on Tuesday 27/11).

Weather

November is one of the rainiest months of the year in Copenhagen. Expect day temperatures between 3 and 7 °C and night temperatures between −1 and 3 °C. The sun will set at 3:57 pm to rise again at 8:09 am.

Social event

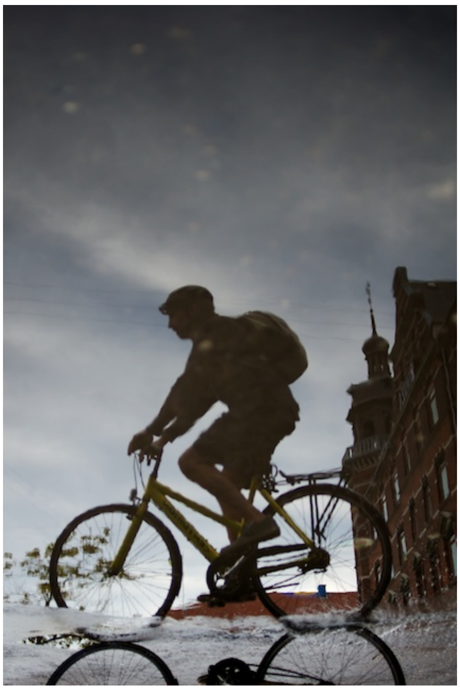
Copenhagen expert Line will guide us through a walking tour of the city's latin quarter on Monday evening. We will start from the meeting venue after the last session at 6:30 pm and the tour will take us all the way to our dinner restaurant, where we can then enjoy a well deserved meal at 8:00 pm. In case the weather is on the chilly side, remember to bring a warm jacket. Umbrellas may prove useful as well, but let's not be too pessimistic!

Meals

A small welcome lunch buffet will be served upon arrival (from 12 pm on Monday) in the Kosmopol Restaurant. Dinner on Monday will be at 8 pm at restaurant "Den Lille Fede", lunch on Tuesday at 1 pm at "Bistro Pastis". If you have specific dietary restrictions, please inform the organizing committee before the meeting.

Programme overview

Monday	Tuesday
Day 1: Afternoon 26	*Day 2: Morning* 27
12:00-12:45 Welcome lunch Buffet in the Kosmopol Restaurant.	**07:30-08:30** Breakfast Breakfast and pick-up at Hotel Alexandra at 8:00.
12:45-13:00 Introduction Kosmopol Auditorium.	**08:30-10:00** Session 3 *Auditory signal processing and virtual environments.* Kosmopol Auditorium.
13:00-15:00 Session 1 *Perceptual effects of hearing-device signal processing.* Kosmopol Auditorium.	**10:00-11:30** Posters B Posters and coffee/tea in the Kosmopol Lobby.
15:00-16:30 Posters A Posters and coffee/tea in the Kosmopol Lobby.	**10:45-11:30** Board meeting ARCHES board members meet in Arboretet (poster session continues).
16:30-18:30 Session 2 *Speech production and recognition in adverse conditions.* Kosmopol Auditorium.	**11:30-12:30** Session 4 *Music-induced hearing loss.* Kosmopol Auditorium.
18:30-20:00 Social event Guided tour of Copenhagen's latin quarter with Line.	**12:30-12:45** Feedback Kosmopol Auditorium.
20:00-22:00 Dinner Dinner at *Den Lille Fede.*	**12:45-14:00** Lunch Lunch at *Bistro Pastis.*



Meeting abstracts

Starred author names are presenter names. Posters will be displayed on both meeting days.

Session 1: Perceptual effects of hearing-device signal processing

Monday, November 26 – 13:00-15:00
Kosmopol Auditorium

13:00

Effects of noise reduction on AM discrimination for normal-hearing and hearing-impaired listeners. D. Timothy Ives* (Ecole Normale Supérieure, 75005, Paris, France), Sridhar Kalluri, Olaf Strelcyk (Starkey Hearing Research Center, Berkeley, CA, 94704, USA), Stanley Sheft (Communication Disorders & Sciences, Rush University Medical Center, Chicago, Illinois, USA), Christian Lorenzi (Ecole Normale Supérieure, 75005, Paris, France).

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.

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.

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.

In conclusion, 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.

13:30

Differences in benefit from a binaural noise reduction algorithm: Effects of hearing loss and cognitive function. Tobias Neher*, Giso Grimm, Christoph Völker, Volker Hohmann, Birger Kollmeier (Medical Physics Group, Carl-von-Ossietzky University, Oldenburg, Germany).

Benefit from hearing aid (HA) signal processing can vary widely, even among listeners with similar audiometric characteristics. So far, little progress has been made in terms of finding ways of tailoring HA fittings better to the needs of the individual. Consequently, HA fittings are typically based on the audiogram only. More recently, a number of studies have identified cognitive function as a potential avenue for more individualization. For example, Gatehouse et al (IJA, 45:153-171) found that amplitude compression interacts with cognitive processing.

It remains unclear, however, how cognitive function impacts listening performance with other types of HA signal processing. Furthermore, for a given algorithm there are typically a number of parameters to adjust, e.g. the signal-to-noise ratio (SNR) at which the algorithm engages or the amount of processing it applies. It is possible that individual differences in auditory and cognitive abilities interact with such parameters. If this were the case, it would be useful to develop fitting rules for such parameter sets.

The purpose of the present study, which at the time of writing is still underway, is to investigate how hearing loss and cognitive function interact with the benefit from a binaural noise reduction (NR) algorithm. To that end, four groups of elderly hearing-impaired listeners are being tested, i.e. groups with either mild or moderate hearing loss and either better or poorer cognitive function. The algorithm under consideration is a binaural coherence-based NR scheme designed to suppress reverberant signal components as well as diffuse background noise. The strength of the processing is varied from inactive to aggressive, and testing is carried out across a range of fixed SNRs. Potential benefit is assessed using a dual-task paradigm, which combines speech recognition in cafeteria noise with a visual reaction time task. Pairwise preference ratings are also being gathered.

The collected data will be presented and discussed.

14:00

Enhanced temporal coding in auditory prostheses can lead to improved sound perception. Raphael Koning*, Jan Wouters (Research Group ExpORL, University of Leuven, Belgium).

Users of auditory prostheses like hearing aids (HAs) or cochlear implants (CIs) suffer from a rapid decrease in speech intelligibility (SI) in adverse listening conditions although their respective speech reception in quiet can be very good.

Recent studies have confirmed that phoneme transitions and rapid changes in temporal and spectral content are the most important parts of the speech signal for SI. The enhanced envelope (EE) strategy has been developed to enhance speech by amplifying the onsets of the speech envelope. The amplification is done in all frequency bands by deriving and introducing additional peak signals at the onsets.

Two groups of listeners participated in the subjective evaluation of the EE algorithm: HA and CI users. The influence of the enhancement of the onsets of the speech envelope was investigated in stationary speech shaped noise (SSN) and with an interfering talker. Keyword correct scores were collected for the Dutch LIST sentences in stationary SSN with six CI users at fixed signal-to-noise ratios (SNRs). The speech reception threshold (SRT) was determined with an adaptive procedure in stationary SSN for HA users and with an interfering talker for CI users. The presentation level of the speech was fixed at 65 dB SPL. For HA users, the presentation level was fitted to the hearing loss with the NAL-rules .

In stationary SSN, both groups of listeners showed an immediate benefit of the EE algorithm in comparison to the unprocessed signal. A SRT improvement around 1.5 to 2.5 dB was obtained for HA users. Also the keyword percent correct scores significantly increased for the CI users corresponding to an SRT improvement of around 2 dB. In the competing talker condition, an SRT improvement of around 3 dB was obtained when the onsets of the target speaker were enhanced. Furthermore, an improvement was obtained when the onsets of the noisy mixture were enhanced. The latter enhanced condition does not require a priori knowledge of the speech signal for the peak signal extraction.

The results suggest that SI can be improved in adverse listening conditions by an enhanced temporal coding of the onsets of the speech envelope in the signal processing path of HAs and CIs.

14:30

Speech enhancement for cochlear implants using a dictionary learning algorithm. Ioanna Avramidou* (ETH Zurich, University Hospital Zurich and Phonak), Norbert Dillier (University Hospital Zurich), Manuela Feilner (Phonak).

Speech recognition in a noisy environment has always been challenging for hearing impaired people. The topic of Speech Enhancement -estimation of the underlying speech component in degraded speech signals- was investigated in relation to Hearing Instruments, with special emphasis on Cochlear Implants.

A machine learning method lies at the core of the algorithm used for Speech Enhancement. This method trains dictionaries as models of signal classes, such as the speech class or a specific noise class. The input signals are represented by (coded on) their corresponding dictionaries through sparse coding. In this work, the performance of the algorithm was optimized with respect to its main parameters, both by objectively measuring the SNR gain achieved after enhancement and by subjectively examining the quality of the output.

In order to examine the algorithm's performance with respect to Cochlear Implants, a Cochlear Implant Simulator was employed. Processing of speech signals with the Simulator has proven to cause to Normal Hearing people a similar impairment in speech understanding to the one observed for Cochlear Implant patients.

The optimized algorithm was evaluated through adaptive SRT tests in the University Hospital of Zurich. Degraded speech files were presented to the test subjects following to being enhanced by the algorithm. The goal was to measure the improvement in speech intelligibility achieved by the introduction of the algorithm and to compare between a sharp and a soft parameterization set. Two subject categories participated in the study, CI patients and NH people. For the latter, the signals were additionally processed with the CI Simulator.

Finally, a modification of the standard algorithm was proposed. There, the algorithm operates in the Wavelet domain instead of the Fourier domain. The aforementioned variation was compared to the standard one in terms of both speech intelligibility improvement and computational cost.

Session 2: Speech production and recognition in adverse conditions

Monday, November 26 – 16:30-18:30
Kosmopol Auditorium

16:30

“Talking in the dips” - Speech production in modulated noise. Ewen N. MacDonald* (Center for Hearing and Speech Sciences, Technical University of Denmark), Stefan Raufer (Institut für Hörtechnik und Audiologie, Jade Hochschule Oldenburg).

The Lombard effect refers to the phenomenon where talkers automatically increase their level of speech in a noisy environment. While many studies have characterized how the Lombard effect influences different measures of speech production (e.g., F0, spectral tilt, etc.), few have investigated the consequences of temporally fluctuating noise. In the present study, 20 talkers produced speech in a variety of noise conditions including both steady-state and amplitude modulated white noise. While listening to noise over headphones, talkers produced randomly generated 5 word sentences presented on a computer monitor. Similar to previous studies, when the noise level was increased, talkers raised the level of their voice and sentence duration increased regardless of whether the noise was amplitude modulated or steady-state. However, the increase in voice level was much smaller then observed in previous studies, particularly in amplitude modulated noise. Importantly, for 2 and 4 Hz amplitude modulated noise conditions, talkers altered the timing of their utterances, reducing the energetic overlap by approximately 2%. However, for the 1 Hz amplitude modulated condition, talkers increased the overlap by approximately 4%. Overall, the results demonstrate that talkers are sensitive to the temporal aspects of noisy environments and will alter their speech accordingly.

17:00

The influence of linguistic skills on speech recognition in noise in normal hearing listeners. Marre W. Kaandorp* (ENT/audiology, VU University Medical Center, Amsterdam, The Netherlands), Annette M. B. de Groot (Faculty of Social and Behavioural Sciences, University of Amsterdam, Amsterdam, The Netherlands), Joost M. Festen, S. Theo Goverts (ENT/audiology, VU University Medical Center, Amsterdam, The Netherlands).

Background: In counselling of hearing-impaired people, judgment of cochlear-implant or hearing-aid candidacy, and evaluation of rehabilitation progress there is need for a more detailed understanding of the factors that influence speech recognition in noise. This study focuses on the role that linguistic skills play in speech recognition in noise. Most important in language comprehension is the process of word recognition. Therefore, we focused on visual measurements of lexical access abilities and vocabulary size. We applied these tests, as well as a set of speech recognition tests, to normal hearing listeners with varying linguistic abilities to investigate how performance on speech recognition in noise is related to lexical word recognition. These data will serve as reference data for experiments in hearing impaired subjects.

Objective: The primary objective of this study was to examine how lexical access abilities and vocabulary size influence speech recognition abilities in noise in normal hearing listeners with varying linguistic abilities.

Method: Speech recognition scores were measured for digit-triplets in steady-state noise and short meaningful sentences in steady-state and fluctuating noise. Lexical access was measured with a lexical decision test (LDT) and with a simple word naming test (WNT). Vocabulary size was measured with a subtest of the Groningen Intelligence Test (GITvs). To introduce variation in linguistic skills, three groups of 24 young normal hearing listeners were included: high-educated natives, low-educated natives, and high-educated non-natives. First, the LDT will be evaluated and correlations with WNT and GITvs will be examined. Secondly, the relation between speech recognition data and linguistic skills will be analysed for the three groups and for individual subjects.

Discussion: This study shows the influence of some linguistic abilities on speech recognition performance in normal hearing listeners with varying linguistic skills. This relation in normal hearing listeners will help the interpretation of future results on the effect of these factors on speech recognition in noise in hearing impaired listeners.

17:30

Temporal-envelope reconstruction for hearing-impaired listeners. Christian Lorenzi*, Nicolas Wallaert (Institut d’Etude de la Cognition, Ecole Normale Supérieure, CNRS), Jayaganesh Swaminathan (Sensory Communication Group, Research Laboratory of Electronics, Massachusetts Institute of Technology).

Recent studies suggest that normal-hearing listeners maintain robust speech intelligibility despite severe degradations of amplitude-modulation (AM) cues, by using temporal-envelope information recovered from broadband frequency-modulation (FM) speech cues at the output of cochlear filters. This psychophysical study aimed to assess whether cochlear damage (and the associated reduction in frequency selectivity) alters this capacity to reconstruct temporal-envelope information from FM.

This was achieved by measuring the ability to identify nonsense VCVCV logatomes processed to degrade AM cues while leaving FM cues intact within three broad frequency bands spanning the frequency range from 65 and 3645 Hz. These bands were presented for identification separately or in combination. In each condition, speech identification was measured for normal-hearing listeners and listeners with mild-to-moderate hearing loss (flat and sloping audiometric configurations).

Hearing-impaired listeners showed significantly poorer identification scores than normal-hearing listeners. However, the deficit shown by hearing-impaired listeners was relatively modest, and peaked for the highest frequency band centered on 2535 Hz. Overall, hearing-impaired data and the results of simulation studies were consistent with a poorer-than-normal ability to reconstruct temporal-envelope information resulting from a broadening of cochlear filters by a factor of 2.

These results indicate that temporal-envelope reconstruction from broadband FM is an important, early auditory mechanism contributing to the robust perception of speech sounds in degraded listening conditions. These results also suggest that most people with mild to moderate cochlear hearing loss can make efficient use of reconstructed envelope cues despite degradations in frequency selectivity. Still, these results suggest that poorer-than-normal frequency selectivity impairs somewhat temporal-envelope reconstruction mechanisms.

18:00

The role of across-frequency envelope processing for speech intelligibility prediction. Alexandre Chabot-Leclerc*, Søren Jørgensen, Torsten Dau (Centre for Applied Hearing Research, Department of Electrical Engineering, Technical University of Denmark).

An investigation of the essential elements of speech intelligibility models is presented. The investigation is based on the concept that intelligibility models can be split into two parts: a preprocessing section, which transforms the speech signal into an internal representation, and a decision metric, which quantifies the effects of processing on the speech intelligibility. The preprocessing analysis presented here contrasts the independent-channel processing, as used in the speech-based envelope power spectrum model [sEPSM; Jørgensen and Dau (2011)]. J. Acoust. Soc. Am. 130, 1475-1487), to the across-channel processing used in the spectro-temporal modulation index [STMI; Elhilali et al. (2003). Speech Comm. 41, 331-348]. The independent-channel processing of the sEPSM extracts the temporal envelope from each channel and then filters each envelope through a modulation filterbank. The across-channel processing of the STMI extracts the spectro-temporal modulation energy of the time-frequency representation of the signal using a two-dimensional filterbank. The decision metric analysis compares the modulation transfer function metric used by the STMI, which computes the reduction of the modulation energy of the processed speech in comparison to the clean speech, to the speech-to-noise envelope power ratio (SNRenv) metric of the sEPSM. While the MTF in the STMI is transferred directly to intelligibility predictions, the SNRenv is related to intelligibility using the concept of an ideal observer. The contributions of each preprocessing and decision metric were evaluated by comparing predictions to data of speech intelligibility under conditions of reverberation, phase jitter and spectral subtraction processing, and by inspecting the internal representation of the models. It is shown that a model combining across-channel preprocessing with the SNRenv metric and the ideal observer can account for the effects of the processing for all three distortions. However, a simple weighting of the across-frequency variance of the modulation power at the output of the (temporal) modulation analysis appears to be sufficient to account for the data, i.e., a joint two-dimensional modulation filterbank might not be required.

Session 3: Auditory signal processing and virtual environments

Tuesday, November 27 – 08:30-10:00
Kosmopol Auditorium

08:30

Timing of the auditory brainstem response to speech is dependent on cochlear origin. Helen Elizabeth Nuttall*, David R Moore, Jessica de Boer (MRC Institute of Hearing Research, Nottingham, UK).

The auditory brainstem response (ABR) is a scalp-recorded evoked potential that reflects the summed firing of many brainstem neurons in response to a sound stimulus. When a speech stimulus is used, the resulting 'speech-ABR' consists of a sequence of peaks that are phase-locked to the envelope of the stimulus. In background noise, the timing precision of the speech-ABR is disrupted and peak latencies increase. The extent of the noise-disruption has been associated with a listener's ability to understand speech in noise.

To date, although a neural mechanism is assumed, very little is known about what causes the disruption of speech-ABR timing. Previous research using simple stimuli has shown that the latency of ABR peaks is strongly dependent on cochlear site of origin, due to the increasing cochlear travel ling wave delay with decreasing frequency. The present study aimed to investigate whether noise-induced latency increases in the speech-ABR could be explained by changes in cochlear origin.

Speech-ABRs to a synthetic [da] stimulus presented at 70 dB SPL were recorded from normally-hearing adults. Intense high-pass noise masking was used to restrict the contribution of selected regions of the cochlea. By subtracting the responses obtained using noise maskers with different high-pass cut off frequencies, 'derived band' responses were obtained that were assumed to reflect contributions from cochlear frequency regions between 0.5-1 kHz, 1-2 kHz, 2-4 kHz and 4-8 kHz.

Preliminary results show that all derived frequency bands made a significant contribution to the speech-ABR, with the largest contribution from the 2-4 kHz region. The response latency increased by 3 ms from the highest (4-8 kHz) to the lowest (0.5-1 kHz) frequency region. These findings suggest that changes in cochlear origin could lead to significant differences in speech-ABR latency. Researchers analysing speech-ABRs in noise should take into account the frequency composition of the masker and its potential effect on cochlear excitation patterns.

09:00

Model-based hearing aids: from psychoacoustics to dynamic compression. Stephan D. Ewert*, Steffen Kortlang, Giso Grimm, Stephan M. A. Ernst, Volker Hohmann (Medizinische Physik, Universität Oldenburg).

Sensorineural hearing loss (HL) like common age-related hearing loss typically results in elevated thresholds and steepened loudness growth with level (recruitment phenomenon). Loudness recruitment typically can be attributed to a loss or dysfunction of the outer hair cells (OHC), while elevated thresholds can be caused by damage of inner hair cells (IHC), too. On the level of the basilar membrane (BM), OHC loss results in a reduced gain for low-level signals, changing the BM input-output function. IHC loss can be interpreted linear damping in the mechano-neuronal transduction process independent of stimulus level accompanied by a loss of temporal coding fidelity. Ewert and Grimm [in Speech Perception and Auditory Disorders, 3rd ISAAR Meeting (2012)] have suggested a model-based, fast-acting dynamic compression algorithm which aims at restoring the normal-hearing BM input-output function in hearing impaired listeners. Additionally, a fitting rule that estimates low-level gain loss from adaptive categorical loudness scaling data and audiometric thresholds based on Jürgens et al. [Hear. Res. 270, 177 (2011)] was suggested. Here, an enhanced version of the compression algorithm is suggested, incorporating a wideband control circuit which reduces gain in broadband stimulus conditions. The circuit also operates on a longer time constant (50 ms), reducing non-linear distortions. The dynamic compression algorithm was evaluated by measuring speech intelligibility in spatial masking conditions, including a bilateral and a binaural version of the algorithm. Additional psychoacoustic experiments are presented with the aim to improve diagnostics of IHC and OHC loss or dysfunction.

09:30

Capturing realistic acoustic scenes using a spherical microphone array. Marton Marschall* (Centre for Applied Hearing Research, Technical University of Denmark), Sylvain Favrot (Department of Speech, Language & Hearing Sciences, Boston University), Jörg Buchholz (National Acoustic Laboratories, Australian Hearing), Torsten Dau (Centre for Applied Hearing Research, Technical University of Denmark).

A systematic investigation of human auditory processing in complex environments, or the "cocktail-party" phenomenon, requires the presentation of realistic and reproducible acoustic scenes around the listener. Such complex scenes may be played back using virtual sound environments, such as the Spacelab facility at the Centre for Applied Hearing Research (CAHR). In order to extend the existing setup to be able to present real-life scenes in addition to simulated ones, a spherical microphone array using mixed-order Ambisonics (MOA) was developed to capture sounds with a high spatial resolution. The goal of the MOA approach was to combine a 3D representation of a certain spherical harmonics order (certain spatial resolution) with a higher-order (higher resolution) horizontal representation. The approach takes advantage of the fact that (i) an increase in resolution in only 2 dimensions may be achieved by a smaller increase in the number of transducers, and that (ii) the properties of human hearing make it desirable to provide a better resolution in the horizontal plane when a limited number of transducers are available. To demonstrate the theoretical advantages and predict the real-life performance of a MOA array, the chosen layout and a regular microphone layout were compared in terms of robustness to noise and microphone characteristic variations. Existing metrics for the performance assessment of arrays needed to be adapted to the mixed-order approach due to different behaviors in the horizontal and vertical directions. The results showed that the expected performance benefits of MOA are maintained under simulated noisy conditions at mid to high frequencies. Performance at lower frequencies could be improved by optimizing the regularization scheme. The spatial scene recordings provided by the array will be used in psychophysical experiments, and can also be utilized for hearing-aid evaluation and mobile phone development.

Session 4: Music-induced hearing loss

Tuesday, November 27 – 11:30-12:30
Kosmopol Auditorium

11:30

Hearing at risk: MIHL. Wouter A. Dreschler*, Monique Leensen, Hiske Helleman, Marya Adluni-Sheikh Rashid (Academic Medical Center Amsterdam).

This paper presents an overview of recent work in our group devoted to Music Induced Hearing Loss (MIHL). Usually, preventive actions are focussed to Noise-Induced Hearing Loss in an occupational setting. However, there seems to be a growing trend that exposure to loud sounds during leisure time may form an equal or even larger risk to hearing than occupational noise exposure.

Study 1 focuses on occupational hearing loss, but with music as the sound source. The results of 241 professional musicians in symphony orchestra illustrate that MIHL is an issue, even in classical music. Reduced function is not only present in the pure-tone audiogram, but it is also expressed in abnormal loudness perception, Oto-Acoustic Emissions, Tinnitus, and Diplacusis. Poorer speech-in-noise thresholds illustrate the consequences for daily-life communication.

Study 2 summarizes the results of an online speech-in-noise test that was adapted to be more sensitive for high-frequency hearing losses. In a sample of 62.782 youngsters (12-24 yr) who voluntarily visited the website, about 4% had poor results and in 28% of the subjects the results were slightly less than normal.

We applied the ISO-1999 (P50) standard to make predictions of potential damage after 10 years of mp3-use and did another online test on critical characteristics of mp3-player use. In a first sample of respondents (study 3 in 100.253 responders) mp3-use as such (isolated from other sources of sound exposure) is a potential risk for about 10% of the mp3-users between 12 and 24 years. In a follow-up study (study4) more details could be collected in a sample of 32,522 young mp3-users.

There is need for an integrated approach in which the risks of different types of sound exposure can be combined in order to counsel the individual sound consumer. The model of Williams (Sydney) offers an elegant starting point to be developed further. Also there is need for a big research project on a European level that addresses the issues of music and noise exposure. We are looking for partners within ARCHES.

12:00

Development of an online questionnaire to investigate long-term music-induced hearing loss. Robert Mackinnon*, Heather Fortnum (NIHR Nottingham Hearing Biomedical Research Unit, Nottingham, UK), Dave Moore (Cincinnati Children's Hospital Medical Center, Cincinnati, OH, US), Marcel Vlaming (NIHR Nottingham Hearing Biomedical Research Unit, Nottingham, UK), Christian Fullgrabe (MRC Institute of Hearing Research, Nottingham, UK).

There is a great deal of concern about the effect on hearing of listening to loud music. While there is clear evidence to support short-term hearing disruption in some circumstances (Zhao, Manchaiah et al. 2010), there is limited evidence to support longer-term or permanent damage, and the evidence that does exist is mixed (Meyer-Bisch 1996; Tambs, Hoffman et al. 2003).

To address this deficit, an online long-term retrospective self-report questionnaire, the Music Exposure Questionnaire, was designed (Mackinnon, Fortnum et al. 2012). The questionnaire contains questions on hearing, otological history, and most importantly asks participants for a life-history of their exposure to loud music.

We anticipate recall of such a life history to be difficult. To facilitate recall, a questionnaire tool called a "calendar instrument" was used. This tool has been shown to improve the quality of data collected in some interviewer-administered retrospective surveys (Belli, Smith et al. 2007). As no studies were identified where a calendar instrument has been used in an online, self-report context, we compared a questionnaire using the instrument to a "standard" questionnaire in a set of two experiments (n=212).

Participants self-rated the questionnaires as being equivalent for difficulty, ease of use, and accuracy of recall ($p < 0.001$). They rated the calendar as being more than moderately helpful. Recalled data was comparably reliable ($p < 0.05$) but more exposure was reported using the calendar ($p < 0.05$), indicating better recall of exposures.

The calendar instrument version of the questionnaire will therefore be used to answer the larger question of whether music-induced hearing loss happens over the long term. This main study will deliver the questionnaire, along with an online hearing test, to 6000 participants in one of the largest surveys of its kind. This will allow us to investigate associations between reduced hearing ability and music-dominant noise exposure.

If associations are found, this work will contribute evidence to support public health advice, and help prevent future generations from acquiring a music-related hearing loss in the first place. Ultimately it is hoped that this work will be extended to larger and more diverse populations via internet delivery.

Belli, R. F., L. M. Smith, et al. (2007). "Methodological comparisons between CATI event history calendar and standardized conventional questionnaire instruments." Public Opinion Quarterly 71(4): 603-622.

Meyer-Bisch, C. (1996). "Epidemiological evaluation of hearing damage related to strongly amplified music (personal cassette players, discotheques, rock concerts)–high-definition audiometric survey on 1364 subjects." Audiology 35(3): 121-142.

Mackinnon, R.C., H. Fortnum, et al. (2012) "An online questionnaire to investigate long term music-induced hearing loss." Presentation given at the BSA conference 2012. International Journal of Audiology (in press).

Tambs, K., H. J. Hoffman, et al. (2003). "Hearing loss induced by noise, ear infections, and head injuries: results from the Nord-Trondelag Hearing Loss Study." International Journal of Audiology 42(2): 89-105.

Zhao, F., V. K. C. Manchaiah, et al. (2010). "Music exposure and hearing disorders: An overview." International Journal of Audiology 49(1): 54-64.

Posters

Monday, November 26 – 15:00-16:30
Tuesday, November 27 – 10:00-11:30
Kosmopol Lobby

P-1

A screening test for air-bone gaps to increase the potential of audiometric screening in non-quiet test environments. Monique Boymans*, Wouter A. Dreschler (Clinical & Experimental Audiology AMC, Amsterdam, The Netherlands).

We like to introduce a system for *direct prescription* by the dispenser in the Netherlands. This is only feasible if the dispenser applies a kind of *triage* in order to assess whether the intervention of an ENT doctor is necessary. Therefore, we need a reliable assessment of the Air-Bone Gap, in order to identify a conductive hearing loss.

EN15927 (2010) provides relatively lax audiometric requirements for the maximum allowable background noise levels in audiometry areas. For many hearing aid candidates with a mild hearing loss (typically 20-30 dB HL for the lower frequencies) *it will be impossible to reliably detect a conductive component* in the hearing loss.

For this purpose we modified the old-fashioned SAL-test revisited (Rainville, 1955; Jerger & Tillman, 1960) in order to avoid the main limitations: occlusion (at low frequencies) and upward spread of masking (at high frequencies). The modified version was called the SAG test (**S**creening on **A**ir-bone **G**aps) and the specificity and sensitivity of the new test was tested in two pilot experiments.

A combination of the SAG test and audiometric screening can provide a safe tool (sensitivity ¿ 90%) for referral to the ENT doctor based on audiological criteria. The triage is not very efficient (low specificity). One of the reasons is the strict criteria for asymmetry. The SAG-test is simple to use and the test can be conducted in non-quiet test rooms (that fulfill the EN 15927, 2010)

P-2

The role of peripheral adaptation and cross-frequency coherence for auditory stream segregation of pure tones. Simon Krogholt Christiansen*, Torsten Dau (Centre for Applied Hearing Research (CAHR), DTU, Denmark).

One of the most extraordinary features of the human auditory system is its ability to perceptually segregate concurrent sounds. This ability is crucial for our ability to use acoustic information in a natural situation, but the underlying mechanisms are still poorly understood. In this study, the computational model of the 'effective' processing of the auditory periphery by Dau et al. (1997) is combined with the conceptual 'temporal coherence analysis' by Elhilali et al. (2009) to create a modelling framework that can predict auditory stream formation. Through this framework, the influence of physiological forward masking on auditory stream segregation is investigated in the classical van Noorden (1975) paradigm. Furthermore, the concept of temporal coherence as a grouping mechanism is investigated, through new experimental data using pure tones with a large frequency separation and varying degrees of asynchrony, and through analysis of the developed model.

P-3

HoerTech master hearing aid development kit. Graham Coleman*, Tobias Herzke (HoerTech gGmbH, Oldenburg), Volker Hohmann (University of Oldenburg).

The Master Hearing Aid (MHA) is a real-time, cross-platform signal processing platform designed for testing and validation of hearing aid algorithms, developed at Hörtech. In the context of a national collaboration project within the Auditory Valley (Oldenburg and Hannover, Germany), we have created a development kit with IDE support for MHA signal processing development and a set of processing blocks ("plug-ins") that cover all basic hearing aid signal processing elements, as well as templates for creating new plug-ins. The IDE support is provided by the open source Qt Creator, and on-line class documentation and tutorials for MHA signal processing development are included.

Available signal processing blocks include STFT, overlap-add, and filter bank; noise suppression, dereverberation, directional microphones, feedback cancellation; equalizer, multi-band dynamic compressor, FIR and IIR filters; resampling, double-buffering.

New work on the Master Hearing Aid and this development kit is supported by a research project on Individualized Hearing Acoustics from the German Research Foundation (DFG).

P-4

Across-lifespan speech perception when audiometric thresholds are in the normal range. Christian Füllgrabe* (MRC Institute of Hearing Research, Nottingham, UK).

Anecdotal evidence suggests that older listeners experience increased listening difficulties in noisy environments; experimental investigations seem to confirm this age-dependent deficit. However, older persons are generally unaware of their peripheral hearing (loss) and many published studies used lax audiometric inclusion criteria. Hence, lower speech intelligibility could, at least partially, be explained by differences in audibility across age groups. Also, previous age comparisons were generally limited to groups of "young" (e.g. = 30 years) and "older" listeners (e.g. = 60 years).

The aims of this still-ongoing, cross-sectional study are to establish if and how speech-in-noise perception declines across adulthood, even when audibility is in the normal range, and to identify potential contributing factors to this putative decline, such as changes in temporal-fine-structure (TFS) processing and cognitive functioning.

So far, 140 volunteers aged 18-90 yrs were recruited, with a least 15 per age decade up to 80 years. All had unilaterally (UNH) or bilaterally normal hearing (BNH), defined as audiometric thresholds =20 dB HL between 125 and 4000 Hz.

Sensitivity to TFS was measured using two psychophysical tests. In the first, UNH and BNH listeners discriminated a monaurally presented harmonic tone complex from an inharmonic tone complex, obtained by shifting all components of the first complex upwards in frequency. Fundamental frequencies (F0) of 91 and 182 Hz were used. The spectral envelope was fixed by applying a filter with a bandwidth of 1F0 and centered on the 11th harmonic (corresponding to 1001 and 2002 Hz, respectively). In the second test, BNH listeners discriminated a diotic 500- or 850-Hz pure tone from the same pure tone with a phase difference between the two ears.

Eight cognitive tests were administered to assess different cognitive functions such as processing speed, executive function, short-term/working memory, verbal fluency, and reasoning.

Speech-perception abilities are currently being assessed for monaurally presented consonants in quiet and in "steady" and amplitude-modulated speech-shaped background noises at different speech-to-noise ratios.

The results show a large inter-individual variability in monaural and binaural TFS processing with average sensitivity declining already from early middle age, and lower performance on most cognitive tests for the older age groups. Speech-identification scores as a function of age and in relationship to TFS sensitivity and cognitive abilities will be presented at the meeting.

Work supported by the Oticon Foundation.

P-5

What makes different ears sound different? Fredrik Gran*, Isabel Schindwolf, Jesper Udesen (GN-Resound).

Virtual sound is a technique, where sound is presented via headphones using HRTF (Head-Related-transfer function) based processing and perceived by the listener as coming from a loudspeaker located within a room. Using a small database of 7 test subjects HRTFs and corresponding headphones to eardrum responses it is thus possible to listen through the outer ears of the database test subjects.

In this study, it is investigated if there is a correlation between a test person's perceptual auditory impression of different ears in the database and the corresponding difference in a selection of spatial cues between the ears. The perceptual impression was evaluated by a virtual sound paired comparison test with reference, implemented in Matlab, where all combinations of different virtual pairs of ears were tested at angles $[0, 40, 90, 130, 180, 230, 270$ and $320]$ degrees. The excitation signal was band limited white noise presented at 67 dB SPL.

The spatial cues under investigation were: Interaural level difference (ILD), interaural time difference (ITD) and monaural spectral difference.

It is found that ILD and ITD are not correlated with the results of the perceptual testing. Only the monaural spectral difference is significantly correlated at 4 out of the 8 presentation angles. This indicates that the most important cue describing the difference between pair of ears is the monaural spectral cue.

P-6

Application of higher-order ambisonics for dynamic acoustic scene rendering in hearing aid evaluations. Giso Grimm*, Jan Heeren, Stephan Ewert, Volker Hohmann (Medical Physics Group, Carl-von-Ossietzky University, Oldenburg, Germany).

Spatial audio reproduction methods with multi-speaker arrays play an increasingly important role in hearing aid evaluation. Higher-Order Ambisonics (HOA) offers a method to create a well-defined three-dimensional sound field at a certain spatial area in the proximity of the speaker array, referred to as sweetspot. However, like all spatial audio reproduction methods, HOA suffers from spatial aliasing at high frequencies, and decoding-strategy-specific perceptual and physical limitations.

The first part of this study assesses the performance of horizontal HOA systems using objective and subjective measures, focussing on the applicability of HOA systems for hearing aid evaluation and hearing research. Technical measures include a comparison of sound-field simulation and measurement, perceptual localization predictions (Dietz et al. Speech Communication vol 53:592-605, 2011), monaural perceptual quality predictions (Huber and Kollmeier, IEEE tasl vol 14:1902-1911, 2006) and the performance of an adaptive directional micro phone algorithm. To rate the subjective localization accuracy, minimum audible angle (MAA) measurements have been performed with normal hearing listeners.

The second part of the present study describes the apparatus that was designed to record and render complex acoustic scenes with moving sources. The focus of this part lies in the scene generation software, which in contrary to conventional spatial recording techniques aims at recording of acoustic point sources and their trajectories in space. Combining many of these point sources with ambient 1rst order Ambisonics background recordings in slightly different temporal or spatial configuration allows the creation of multiple equivalent acoustic scenes from the same acoustic material.

Methods and first experiences with a prototype system are presented and discussed.

P-7

Effects of spontaneous otoacoustic emissions on frequency discrimination. Rói Hansen*, Sébastien Santurette (Centre for Applied Hearing Research, Technical University of Denmark), Sarah Verhulst (Center for Computational Neuroscience and Neural Technology, Boston University), Torsten Dau (Centre for Applied Hearing Research, Technical University of Denmark).

Spontaneous otoacoustic emissions (SOAEs) are low-level pure-tone like sounds generated by the cochlea that can be measured in the ear canal. While SOAEs appear to have no practical function, they have been shown to affect pitch perception. Specifically, frequency difference limens (DLFs) have been found to systematically improve near an SOAE, both in the ipsi- and contralateral ear, suggesting an effect of central origin due to constant stimulation of specific frequency channels by the SOAE throughout life.

However, DLF measurements rely on the presentation of external pure tones, with which SOAEs are known to interact. SOAEs typically become synchronized with an external pure tone of neighboring frequency, a phenomenon known as "entrainment". An SOAE is entrained when the external tone is close enough in frequency, while a beating pattern between the SOAE and the tone is observed as the tone moves further away from the SOAE.

In the present study, it was hypothesized that DLFs follow the entrainment pattern of SOAEs, such that frequency discrimination thresholds are higher when the SOAE is beating with the external tone than when it is entrained by it. It was also investigated whether such a pattern was observed in the contralateral ear. If not, this would suggest that the improvement in DLF near an SOAE is of peripheral rather than central origin.

Six subjects with strong SOAEs in the ipsilateral ear and no measurable contralateral SOAE were selected. The frequency and level of the SOAEs were identified and their entrainment patterns obtained by presentation of a sweep around the SOAE frequency. Hearing thresholds (HTs) were measured at the SOAE frequency and the entrainment pattern corresponding to 10 dB SL was identified. A total of 9 suitable test frequencies covering the entrainment and beating regions were then determined. HTs at these frequencies were obtained and the DLFs measured ipsi- and contralaterally at 10 dB SL. The presence of entrainment at the tested frequencies was evaluated using a spectral method.

DLFs were found to systematically improve in the entrainment region, while they worsened when beating occurred, and improved again for test frequencies further away from the SOAE. This pattern suggests a strong correlation between the way SOAEs interact with external tones and the observed DLFs. No improvement in DLF near the SOAE frequency was found in the contralateral ear of any of the subjects, strongly suggesting that the effect is of peripheral, rather than central, origin.

P-8

Measuring listening effort with reaction time to digits in noise. Maj Haverkate*, Rolph Houben, Inge Brons, Maaike van Doorn-Bierman, Wouter A. Dreschler (Academic Medical Center Amsterdam, Dept. of Clinical and Experimental Audiology).

Background: Audiological rehabilitation can be beneficial for speech communication even if it does not have a measurable effect on speech intelligibility scores. For example, single-channel noise reduction is often preferred by hearing-aid wearers, while it does not improve speech scores. We hypothesized that this is caused by an improvement in listening effort.

Unfortunately, there is no easy method for the measurement of listening effort. The purpose of the current study is to design an easy, clinically feasible measurement of the reaction time after spoken digits in noise.

We hypothesized that reaction time can be used to measure listening effort at signal to noise ratios (SNRs) that are high enough to render the speech completely intelligible. Additionally we want to determine if the reaction time to spoken digits can be improved by noise reduction processing.

Methods: We added various amounts of stationary noise to spoken digit triplets to obtain different SNRs (-5 dB, 0 dB, 5 dB, quiet). In addition to these "nprocessed" speech-in-noise stimuli we also included stimuli that were processed with two noise reduction schemes (Ideal Binary Masking and a Minimum Mean Square Estimator). All stimuli were used in a reaction time experiment with two different tasks. In the first task, participants had to quickly identify the last digit of a triplet ('identification'). In the second task they had to quickly add the first and the last digit ('arithmetic'). Twelve normal-hearing subjects participated in this study.

Results: Higher SNRs correspond to smaller response times for both the identification and the arithmetic task. The response time on the arithmetic task was more influenced by the noise than the response time on the identification task, but the arithmetic task led to a higher variance. Experiments on the effects of noise reduction on reaction time are still ongoing. Preliminary results will be presented at the conference.

Conclusions: The significant effect of SNR on reaction time implies that it is possible to measure listening effort with spoken digits at SNRs at which speech is highly intelligible. The optimal task may depend on the SNR that is of interest. A potential audiological application is the evaluation of noise reduction and data investigating this application are currently being gathered. If positive, the listening effort test has the potential to fill a gap in the evaluation methods of assistive hearing devices.

P-9

A multi-resolution envelope-power based model for speech intelligibility. Søren Jørgensen* (Centre for Applied Hearing Research, DTU), Stephan Ewert (Medical Physics Section, Carl von Ossietzky-Universität Oldenburg), Torsten Dau (Centre for Applied Hearing Research, DTU).

Jørgensen and Dau [(2011). J. Acoust. Soc. Am. 130, 1475-1487] proposed the speech-based envelope power spectrum model (sEPSM) in an attempt to overcome the limitations of the classical speech transmission index (STI) and speech intelligibility index (SII) in conditions with nonlinearly processed speech. Instead of considering the reduction of the temporal modulation energy as the intelligibility metric, as assumed in the STI, the sEPSM applies the signal-to-noise ratio in the envelope domain (SNRenv). This metric was shown to be the key for predicting the intelligibility of reverberant speech as well as noisy speech processed by spectral subtraction. However, the sEPSM is limited to conditions with stationary interferers due to the long-term estimation of the envelope power. Thus, it cannot account for the greater intelligibility typically obtained in conditions with fluctuating interferers relative to conditions with stationary interferers, denoted as speech masking release. Here, a multi-resolution version of the sEPSM is presented where the SNRenv is estimated in temporal segments with a modulation-filter dependent duration. The model is evaluated in three categories of interferers and distortions that represent different challenges: (i) speech mixed with three types of stationary noise with widely different spectral characteristic, evaluating the model's ability to account for differences in spectral masking; (ii) speech mixed with five types of fluctuating noise with very different temporal structure, evaluating the model's predictions of speech masking release; and (iii) two conditions with processed noisy speech in the form of reverberation and spectral subtraction, testing the model with respect to convolution and nonlinear distortions in the processed speech. The model accounts well for the intelligibility observed for the stationary and non-stationary interferers, as well as the distorted noisy speech, demonstrating further that the SNRenv is a powerful metric for speech intelligibility prediction.

P-10

Perceptual testing with virtual sound. Jesper Udesen*, Fredrik Gran (GN-Resound, Denmark).

Virtual sound is a well established technique where external sound sources up to approximately 6 kHz can be reproduced with headphones. If the technique is based on the users own head related transfer functions (HRTFs) it is possible to create a complex realistic virtual sound environment, using only a PC and a pair of headphones.

We present the first test results based on virtual sound where hearing aid algorithms can be tested in a virtual hearing-in-noise-test (HINT) environment. The method is based on measurements of horizontal plane HRTFs which takes approximately 20 minutes for each test subject when the angular resolution is 10 degrees.

First, it is tested if the virtual HINT has the same performance in terms of speech-reception-threshold (SRT) values as a normal HINT using real speakers. Four speakers at $[-90, 0, 90, 180]$ degrees are used where the center speaker at 0 degrees is transmitting speech sentences from the DanishHINT developed at the Technical University of Denmark. It is found that the real HINT and the virtual HINT are not significantly different in a statistic sense and that the deviation in mean performance is on the order of 1 dB. Furthermore, it is found that the lack of visual cues (there are no speakers present) in the virtual HINT, does not affect performance more than 1 dB.

Secondly, the virtual HINT is repeated on three normal hearing subjects using either: a) the test subjects natural hearing (personal HRTFs), b) a virtual BTE device in omni mode (only the front microphone), c) a virtual BTE device in directionality mode. The virtual HINT is extended to also test for target speech at $[0, 90, 180]$ degrees.

The corresponding mean SRT values with standard deviations are: a) -4.0 ± 1.2 dB, b) -3.2 ± 1.9 dB, c) -3.2 ± 1.4 dB. These results are not statistically different but they suggest that the test persons perform better in a noisy environment when using their own natural hearing.

List of participants

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Map

Acknowledgments

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