Impact of beamforming algorithms on speech perception in cochlear implant users in a moving noise source condition

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ABSTRACT

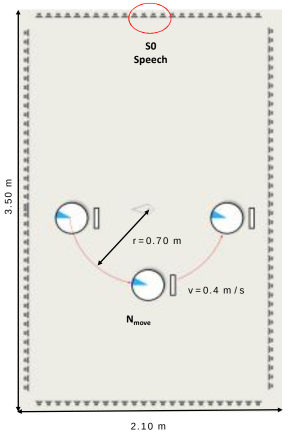
The assessment of speech perception in noise in hearing aid or cochlear implant (CI) recipients in clinical audiological diagnostics is typically carried out in setups containing only a few loudspeakers. In order to determine the impact of advanced signal processing algorithms in everyday situations, a more sophisticated measurement setup is demanded.

A multi-channel sound reproduction system containing 128 loudspeakers was developed and realized in the anechoic chamber of our department [1]. The sound reproduction method 'Wave Field Synthesis' (WFS, [2]) is used to assess speech perception of CI users in realistic auditory environments.

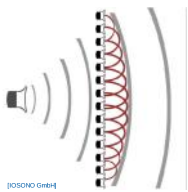
MATERIAL AND METHODS

- 12 normal hearing controls and 14 CI users (7 bimodal and 7 bilateral, Cochlear CP810 speech processors)
- CI users tested with everyday setting and static (ZOOM) and dynamic (BEAM) beamformer
- adaptive measurement of speech reception threshold (SRT) with Oldenburg Sentence Test (OLSA, [3]) in a situation with one moving noise source
- quasi-continuous Oldenburg Noise (Ol-Noise, [3]) and modulated Fastl-Noise [4]
- moving noise sources are virtual sound sources generated by WFS

Measurement conditions



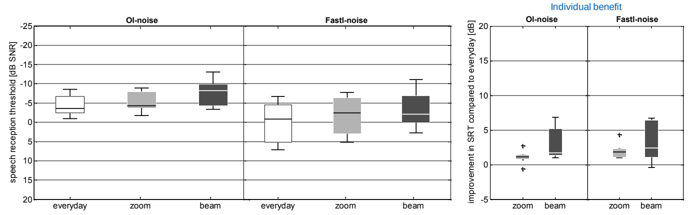
Wave Field Synthesis



The sound field created of a virtual sound source (primary source) is reproduced by interference (level and delay adjustment) of secondary sources with infinitesimal spacing.

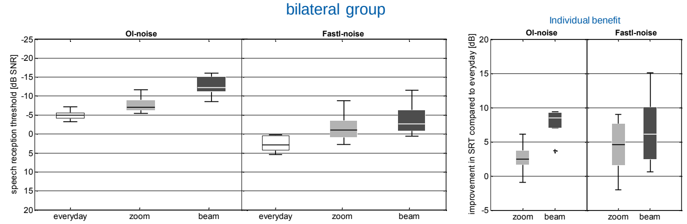
RESULTS

bimodal group



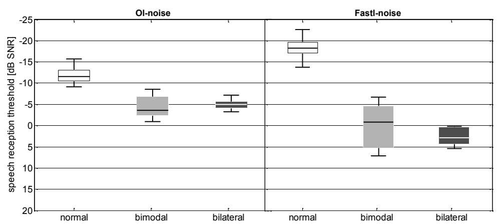
For both noise conditions the use of beamforming algorithms lowered SRT. Because of the high individual variation in hearing within the bimodal group and the little sample size only a tendency was found. The adaptive algorithm BEAM gained the maximum benefit of 7 dB. There is no individual benefit of BEAM compared to ZOOM in both conditions but a tendency of lower SRT due to the higher directivity of the beam algorithm.

bilateral group



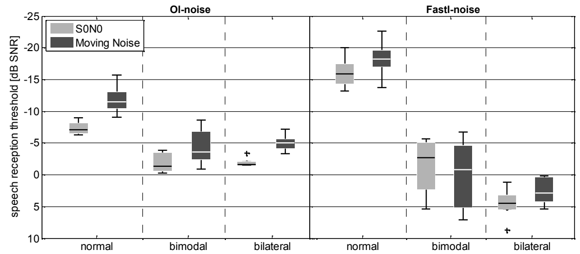
The beamforming algorithms reduced SRT significantly in both noise conditions. The adaptive algorithm BEAM gained a maximum benefit of 7 dB. For Ol-noise, subjects' individual benefit of BEAM is significantly higher than for ZOOM.

Moving noise source, group comparison



The normal hearing control group outperformed both CI groups significantly. For Fastl-Noise, SRT of the bilateral CI group considerably increased (8 dB) whereas the normal hearing group even showed a decrement of SRT of 5.5 dB. The patients of the bimodal group showed lower SRTs in Fastl-Noise than the bilateral group due to their residual acoustic hearing and the ability for glimpsing into the gaps of the time modulated noise.

Spatial release from masking



Improved speech perception due to spatial sound separation is often referred to as 'Spatial Release from Masking (SRM)'. In the Ol-noise condition, the normal hearing group and the bilateral CI group show a significant SRM. However, there is also a tendency of SRM in the bimodal group. In the amplitude-modulated noise condition, only the control group showed a significant SRM whereas the SRT of the bimodal group is even deteriorated by the moving noise source.

SUMMARY

WFS can serve as a sophisticated tool to simulate arbitrary situations of everyday life to measure the technical characteristics of adaptive strategies or to assess patients benefit in defined complex auditory environments.

The beneficial effect of source separation decreased in a amplitude modulated noise condition.

The implementation of adaptive beamforming strategies provides additional benefit in listening situations with a single moving noise source in the rear. A maximum increment of 7 dB in speech perception (dependent on noise condition) was measured.

Advanced listening conditions are necessary in order to evaluate the benefit of recently developed signal processing algorithms in hearing aids or cochlear implant processors.

References

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