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The effect of a linked bilateral noise reduction processing on speech in noise performance

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Directional processing already provides tangible noise reduction benefits in hearing aids but further improvement is needed for hearing-impaired listeners to communicate as effectively as normal-hearing listeners in noisy environments. The objective of this study was to investigate if a binaurally linked beamformer could further improve the signal-to-noise ratio (SNR). Speech reception thresholds (SRT) and spatial perception were compared for bilaterally fitted cardioid microphones and two binaurally linked beamformer processing conditions; 1) a single audio stream output to the two ears, and 2) two audio stream outputs which preserved spatial cues. 10 normal-hearing and 22 hearing-impaired listeners were recruited for this study. The strategies were implemented on a real-time PC processing platform, wired to a pair of behind-the-ear devices via a sound interface. A speech-in-noise test was administered using the Bamford-Kowal-Bench (BKB) sentences targeting the SNR for which 75% correct keywords were identified in spatially separated multi-talker babble noise and room reverberation. The SNR level at which the listeners acquired 95% intelligibility from continues speech discourse material, using a male and a female talker, was also obtained. Sound amplification was provided according to NAL-NL2. Both beamformer conditions improved the SRTs relative to the conventional cardioids, but by a greater degree for the hearing-impaired listeners, and more convincingly at the higher SRTs.

INTRODUCTION

The understanding of speech in noisy listening situations is extremely challenging for hearing-impaired (HI) listeners. Assuming speech levels that are typical in environments with different noise levels (Pearsons *et al*., 1977), and assuming a typical noise spectrum (Keidser, 1995) speech intelligibility index calculations suggest that HI listeners with moderate losses (averaging 50 dB HL) experience a maximum of 50% intelligibility in moderate background noise levels (Dillon, H. 2010). The electro-acoustic amplification provided by hearing aids result in 90% intelligibility in background noise levels not exceeding 60 dBA. However at background noise levels greater than 70 dBA, electro-acoustic amplification provides no more than 10% intelligibility improvement over the unaided ear. When

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hearing aids include directional microphones the intelligibility thresholds at 70 dBA background noise levels are increased by 10% over electro-acoustic amplification alone. Although this improvement is significant and tangible for hearing aid users, further improvement is required for HI listeners to achieve close to NH performance. Background noise levels of 70 dBA and higher are common in our everyday life and thus there is a scope for improving the performance for HI listeners. The aim of this study was to determine if binaural beamformer (BBF) processing can provide a better alternative to more conventional directional microphones featured in modern hearing aids. The processing methods were evaluated at low and high background noise levels in a simulated complex listening situation with multiple competing talkers and room reverberation. The measures of intelligibility and sound quality are discussed herein.

Directional microphones

In hearing aids, directional microphones, sometimes referred to as 1st order microphone arrays, provide directionality by combining the output signals from two spatially separated omni-directional microphone port locations, typically separated by no more than 2 cm. When an internal delay is introduced to one of the microphone output signals and the two microphone signals are combined, then it is possible to produce a directional output signal where different delays result in different directional responses (see reviews by Powers and Hamacher, 2002). In this study, a delay equal to the maximum inter-microphone delay was selected to achieve a sensitivity response commonly referred to as cardioid. Although greater directivities are possible by selecting shorter delays, e.g. hyper-cardioids and super-cardioids, in practice, and in complex listening situations, the directivities achieved are not that different for different delays due to head shadowing and other diffraction effects preventing optimal polar pattern formations. In addition, the directional benefit is not always ideal due to reverberations. The directional benefit of cardioid microphones, and microphones with higher directivities, averages to about 3 dB signal-to-noise ratio (SNR) in real listening situations (Soede, 1990; Powers and Hamacher, 2002; and Kates, 2008). Because directional processing is commonly found in most modern hearing aids, this directional processing condition was used as the reference test condition in this study.

Binaural beamformer

Many binaural beamformer schemes have been proposed in the literature (e.g. Soede, 1990; Markides, 1997; Brandstein and Ward, 2001; and Baptise, 2004). The binaural beamformer (BBF) examined in this study is based on the scheme proposed by Mejia and Dillon (2010), which relates to multi-channel noise suppression algorithms. The scheme works by estimating the acoustic similarity between the left and right directional microphone output signals, and combining these outputs proportionally to their similarities. The block diagram for this algorithm is shown in Fig. 1. In the figure, the output from a pair of cardioid microphones, located on each side of the head, is altered with left and right filters, shown as WL and WR. These filters are estimated by computing the ratio between the time-averaged cross-power between left and right microphone output signals and the power from each

individual directional microphone output. When the power ratio is close to one, the sound is assumed to emanate from the mid-line direction, whereas when the power ratio is lower than one, the sound is assumed to emanate from an off-axis direction. A simple rule is used to suppress only those sound sources at off-axis directions. When this rule is applied to narrow band signals over very short periods of time, an output signal can be reconstructed, which contains a significant portion of sounds emanating from the mid-line direction while significantly suppressing all other sounds. This technique assumes that the desired sound source emanates from a known direction, i.e. mid-line. Although the sound of interest does not always emanate from the mid-line direction; i.e. directly in front of the listener, in practice people do tend to turn their heads to directly face the sound source of interest. This behaviour not only overcomes reduced directionality caused by head shadowing effects (Blauert, 1997), but produces visual cues such as lip reading and face expression cues which are known to assist listeners in speech communication.

Preserving spatial cues

Although BBF processing provides an ideal framework for SNR improvements, in real life applications, BBF processing may also handicap speech communication in noise by disabling the ability of listeners to localise sounds in the acoustic scene (Blauert 1997, 2005). In order to resolve this problem, Mejia *et al.,* (2006), proposed a novel technique which employs the precedence effect, or acoustic suppression of early reflections, to enhance the spatial hearing experience of listeners through BBF processing. As shown in Fig. 1, the output from an ideal BBF is delayed and combined with scaled down (i.e. by 0.2) versions of the microphone output signals from each side of the head. Assuming that the proportion of noise present in the BBF output signal is small, then the scaled down noise sounds present in the left and right directional outputs will precede the similar noise present in the ideal BBF output. As a result of the onset dominance of sounds, listeners perceive the noise present in the left and right directional outputs, hence re-inserting an apparent localisation of all competing noise present in the acoustic scene, and as a trade-off slightly sacrificing the improved SNR achieved with the BBF. This technique, the third included in the examination, is hereafter referred to as the binaural beamformer with preserved spatial cues (BBF-SPC).

METHODS

Subjects

10 NH and 22 HI listeners participated in this study. The average age for the NH group was 35 years, whereas the average age for the HI group was 72 years. The HI listeners all had symmetrical sensorineural hearing loss that averaged 43 dB HL.

Simulated listening situation

Target and noise were presented via eight loudspeakers located in a circle each at a distance of 2 m from the listener situated at the circle center. The front loudspeaker reproduced the target sound with either Bamford-Kowal-Bench (BKB)-like sentences (Bench *et al*., 1979) or continues discourse. The noise sounds included 4 or 16 talkers. In the 4-talker situation, all talkers were spatially separated and presented through the loudspeakers positioned at ±45 degrees and at ±135 degrees from the target sound. During testing, the 4 individual talkers randomly switched between these 4 loudspeakers every 3 secs. In the 16-talker situation, each of 7 loudspeakers presented at least 2 talkers and outmost 3 talkers at any given time. Again, the combination and number of talkers varied randomly between loudspeakers, changing every 3 seconds. The test room (5 x 6 x 2 m) had a T_{60} of 0.4 seconds, corresponding to the acoustics of an average lounge room.

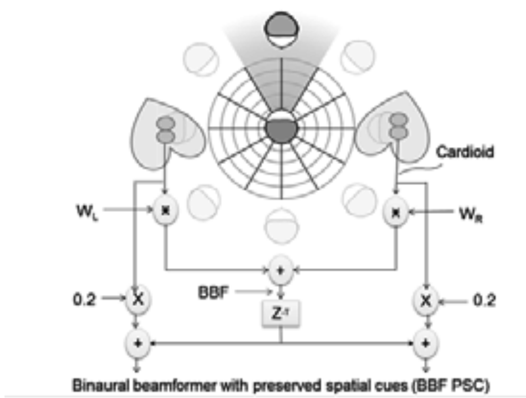

Fig. 1: An overview of the test setup consisting of eight loudspeakers, with the listener situated in the centre facing the target loudspeaker. The figure also illustrates the block diagram for the binaural beamformer schemes evaluated in the study, refer to text for description.

Processing methods

The three processing methods examined were described earlier and were referred to as the cardioid, the BBF, and the BBF-SPC method. The processing methods were implemented in a master hearing aid operating in real-time mode which also included a hearing aid processing module based on dynamic multi-channel compression, which was programmed according to the NAL-NL2 prescription (Keidser *et al*., 2011). The master hearing aid hardware comprised of a laptop (TOSHIBA), a multi-channel audio interface (RME Multiface II), and a pair of behind-the-ear (BTE) casings fitted with two omni-directional microphones and receivers directly wired to the audio interface via a one-meter cable. The separation between the microphones on each BTE unit was 1.2cm.

Intelligibility task

The speech reception thresholds (SRT) were measured at the 95% and 75% points. For the 95% correct estimation, listeners were presented with continuous discourse twice using a male and a female talker. The discourse material was extracted from a compilation of speech and noise sounds available from the National Acoustic laboratories for hearing aid evaluation (Keidser *et al*., 2002). The SRT scores were obtained for the 4-talker noise condition only, where the noise level was fixed to 60

dB SPL. Listeners were asked to adjust the level, using a wireless keypad with 'up' and 'down' controls that enabled them to increase and decrease the speech level, until 95%, of the story was intelligible. For the 75% correct response, listeners were presented with BKB balanced sentences in the 4- and 16-talker background noises, where the noise level was also fixed to 60 dB SPL. To obtain an SRT-SNR score at 75% correct, an automatic procedure and 32 balanced BKB sentences were used. The automation method comprised of presenting one sentence, and increasing or decreasing the sound level of the target depending on the proportion of morphemically correct items identified by the listener. The estimation was improved by progressively decreasing the step size of the sound level proportionally to the number of reversals detected.

Overall preference task

Immediately following the intelligibility test, subjects were asked to select the overall preferred processing of the three processing conditions when listening to both male and female continuous discourse in the 4- and 16-talker background noises at two SNRs. The preference task for the NH group was based on 0 and -5 dB input SNR, while the HI group was presented with 0 and +5 dB free field SNR. These SNRs produced close to 100% speech intelligibility, on average, for each group. In addition to their individual preference, subjects were encouraged to briefly and freely describe the criteria they used for their individual preferences.

RESULTS

Figure 2 shows the average 75% correct SRT-SNRs for each group. A linear analysis of variance showed that at this lower percentage point there was a statistical significant interaction between background noise conditions, hearing loss and processing conditions [SS 77, DF 2, MS 39, F 25, p 0.0000]. A post-hoc test revealed that the HI group acquired a significant benefit from the BBF in the 4-talker background noise but no significant difference was observed in the 16-talker condition. For the NH group, the scores were not significantly different between processing conditions in either noise condition.

Figure 3 shows the average 95% correct SRT-SNRs for each group. For each group, only the performance with the beamformer scheme that produced the best result is shown, which was the BBF-PSC processing for the NH listeners and the BBF processing for the HI listeners. A linear analysis of variance showed a significant difference between processing conditions [$p < 0.001$]. The mean value for BBF-PSC was 2.2 dB better than cardioid for the NH group. The mean value SRT-SNR benefit for the BBF was 2.6 dB better than cardioid for the HI group. A within group comparison suggested that the responses for the unaided NH group were not significantly different than the responses from the HI group aided with BBF processing. For both groups the performance with the other beamformer scheme fell somewhere in between the performances with the cardioid and the better beamformer scheme.

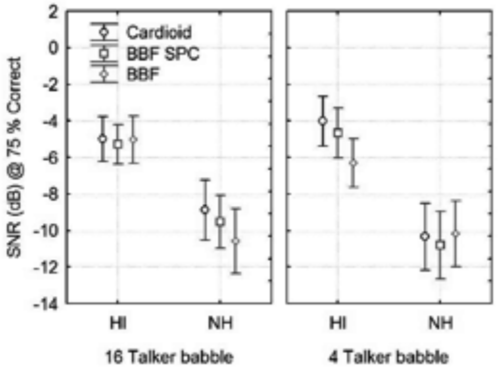

Fig. 2: The average SNR scores at 75% correct responses for normal hearing (NH = 10) and hearing-impaired (HI = 22) listeners. The scores are shown for the the cardioid, BBF SPC and BBF processing conditions in each of the16-talker and 4-talker babble noises. The bars show the 95% confidence intervals.

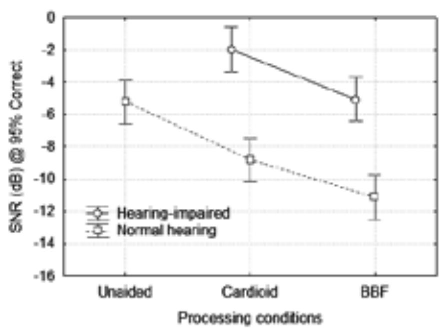

Fig. 3: The average SNR scores at 95% correct responses for normal-hearing (NH = 10) and hearing-impaired (HI = 22) listeners. The scores are shown for the 4 talker babble listening situation. The unaided condition is shown for the normal-hearing listeners only; the cardioid condition is shown for both hearing groups; and the best mode of binaural processing condition is shown for each group, which were the BBF-SPC for the normal-hearing group and the BBF for the hearing-impaired group. The bars show the 95% confidence intervals.

Figure 4 shows the overall processing preference by each group of listeners. Using a Friedman two-way analysis, it was concluded that for the NH group at least one processing condition was significantly different than the two others [$p < 0.001$]. Overall, for the NH group, the BBF-SPC was the most frequently preferred choice, whereas for the HI group there was a noticeable, but not significant, preference for the BFF processing mode. This suggested that for the NH group the spatial naturalness of the BBF-SPC outweighed the SNR benefits (as shown in Fig 2) of the BBF processing, whereas for the HI group the SNR benefits provided by the BFF

processing mode outweighed the possible preservation of localisation cues. In other words, the trade off in preserving the localisation cues for a lower SNR was in fact a detrimental factor for the HI group.

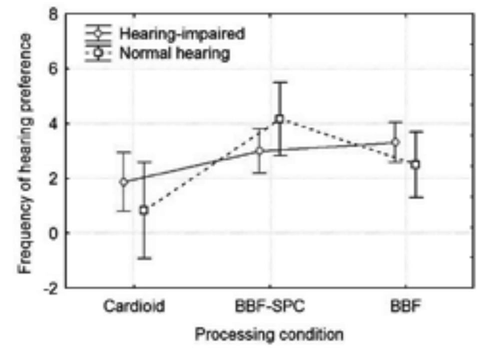

Fig. 4: The average number of times the two groups of listeners showed a preference for the cardioid directional, BBF PSC and BBF processing schemes. While the SNR was adjusted to 0 and -5 dB for the normal-hearing group, the SNR was adjusted to 0 and +5 dB for the hearing-impaired group.

Why was the SNR benefit limited to the 95% intelligibility threshold levels? Based on subjective descriptions noted during the overall preference assessment it is suspected that even though the BBF provides a significant output SNR benefit, the noticeable distortions present in the output signal at lower SNR levels outweighed the overall intelligibility benefit acquired by listeners. In order words, despite the superior output SNR produced by the BBF processing condition over the cardioid condition, the degradation of sound quality was of greater significance than the SNR improvement for the intelligibility task.

SUMMARY

In this study, the BBF processing methods were shown to provide a considerable SNR improvement of 2.2 dB for NH listeners, and 2.6 dB for HI listeners in the 4-talker noise condition at 95% correct responses, but no consistent improvements were observed at lower SRT-SNR threshold levels. Only HI listeners showed a 2 dB advantage with the BBF in the 4-talker noise condition. For the NH listeners, the improvements were greatest with the BBF-SPC processing condition, whereas for the HI group the improvements were greatest with the BBF processing condition. This suggests that while NH listeners take significant advantage of the preserved localisation cues, HI listeners benefit the most from the greater output SNR provided by the BBF without further processing. Overall, NH listeners also had a significant preference for the BBF-SPC, and there was a small, but tangible, preference for the BBF by the HI group. However, BBF-SPC and the cardioid processing conditions were equally preferred by the HI group. Finally, at 95% correct threshold levels, the HI group of listeners acquired similar SRT-SNR score levels as observed by the unaided NH group. Therefore we conclude that although further work is needed to improve processing at even lower input SNR levels, the benefits from BBF processing can significantly assist HI listeners to communicate more effectively in the kind of complex listening situations examined in this study.

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Perceptual comparison of noise reduction in hearing aids

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Knowledge on perceptual consequences of single-microphone noise reduction in hearing aids is limited. We developed and evaluated a filtering method that allowed us to directly compare noise reduction systems from different hearing aids. Using this method, we compared noise reduction from four different hearing aids in a paired-comparisons design. Preference strength of our normal hearing subjects appeared to differ between noise reduction systems as well as between SNRs. Both factors are relevant for the interpretation of previous noise-reduction studies as well as for hearing-aid selection and fine-tuning.

INTRODUCTION

Most modern hearing aids use single-microphone noise reduction to increase listening comfort in noisy environments. Unfortunately, details about the properties of noise reduction in hearing aids are rarely provided. Furthermore, there is limited knowledge about possible benefits of noise reduction. Some studies reported a clear preference of listeners for noise reduction *on* over *off* (Boymans and Dreschler 2000; Ricketts and Hornsby 2005). However, other studies could not confirm such positive effects (Alcantara *et al*. 2003; Bentler *et al*. 2008). Each study compared noise reduction on and off *within* one type of hearing aid. Differences in noise reduction *between* hearing aids may therefore contribute to the diverging results. Thus, there is a need for a method to directly compare different hearing-aid noise-reduction systems to each other, without the dominating effects of other hearing-aid characteristics (e.g. frequency-dependent gain). This led to our first research question:

1. Can we remove the perceptual differences between recordings from different hearing aids, so that they are perceptually equal if noise reduction is turned off?

We designed and evaluated a filter method to remove the perceptual differences. This was described in detail in Houben *et al*. (2011) and summarized here as Experiment 1. The method allowed us to do a paired-comparisons experiment (Experiment 2) in order to answer the following question:

2. Do normal hearing subjects have preference for
   (a) noise reduction *on* over noise reduction *off within* a hearing aid?
   (b) noise reduction from one hearing aid over noise reduction from another?

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