Influence of a remote microphone on localization with hearing aids

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When used with hearing aids (HA), the addition of a remote microphone (RM) may alter the spatial perception of the listener. First, the RM signal is presented diotically from the HAs. Second, the processing in the HA often delays the RM signal relative to the HA microphone signals. Finally, the level of the RM signal is independent of the distance from the RM to HA. The present study investigated localization performance of 15 normal-hearing and 9 hearing-impaired listeners under conditions simulating the use of an RM with a behind the ear (BTE) HA. Minimum audible angle discrimination around an average angle of 45° was measured for three sets of relative gains and seven sets of relative delays for a total 21 conditions. In addition, a condition with just the simulated BTE HA signals was tested. Overall, for both groups, minimum audible angle discrimination was best when the relative RM gain was small (-3 and -6 dB) and the delay was approximately 10-20 ms. Under these conditions, localization performance approached the level obtained in the BTE HA only condition.

INTRODUCTION

Listening in background noise and/or reverberant environments can be particularly difficult for hearing-impaired (HI) listeners. In some situations, it is possible for an HI listener to use a remote microphone (RM) positioned close to the talker and have the signal streamed wirelessly to his or her HA (Ross and Giolas, 1971). Relative to the microphones in the HA, the RM receives a better quality signal from talker (i.e., higher signal-to-noise ratio and/or direct-to-reverberant ratio of the talker), which can improve intelligibility (Hawkins, 1984; Nábělek and Donahue, 1986; Boothroyd, 2004) and possibly reduce listening effort. However, the RM signal is mono and mixed diotically with bilateral HAs. If presented on its own, the RM signal should result in the perceived location of the talker being internalized in the head of the HI listener. Thus, the use of an RM may improve intelligibility but may decrease an HI listener's ability to localize the talker.

For cases where a single RM is in use (e.g., teacher in a classroom, listening to a speaker in an auditorium, etc.) localization may be only a minor issue as the HI

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listener may not need to switch his/her focus away from the RM signal. However, in situations where multiple RMs are in use or multiple talkers are sharing the same RM, judging the location of the different talkers via acoustic cues may be advantageous in keeping up with the conversation.

There are two parameters when mixing the RM signal in the HA: the relative gain and the relative delay of the RM signal. Guidelines for audiologists regarding the relative gain fitting for children have been established and suggest a goal of "Transparency", by setting the gain of the RM to equal that from HA when presented with a 65 dB SPL signal (American Academy of Audiology, 2011). However, to our knowledge, no guidelines for RM delay have been established. Instead, the delay is usually determined by the the digital communication protocol used to wirelessly stream the RM signal to the HA.

The goal of the present study was to investigate the effects of relative gain and delay on spatial perception by comparing minimum audible angle (MAA) thresholds for a source with an incident angle of $45°$ in conditions that simulated using an RM with an ideal behind-the-ear (BTE) hearing aid.

METHODS

Spatialized stimuli were created through acoustic recordings in an anechoic room using two head and torso simulators (HATS, Brüel & Kjær 4128C) with omnidirectional microphones, which were positioned in locations corresponding to an RM and two BTE HAs. The recorded stimuli were then used in the listening test and were presented via headphones in a sound booth.

Stimuli

Recordings of 15 short Danish sentences spoken by a female talker were played back via a "talking" HATS in an anechoic room. These sentences were taken from the corpus created for Sørensen *et al.* (2017).

The speech signals were played back by the "talking" HATS, which was equipped with a mouth simulator. Both the "talking" and "listening" HATS were placed at an equal height with a distance of 97 cm from mouth to mouth and 118 cm from ear to ear (see Fig 1). The listening HATS was placed on a Brüel & Kjær Type 9640 turntable in order to record different angles between the talking and listening HATS.

To simulate an RM, a DPA 4060 omnidirectional microphone was positioned on the chest of the talking HATS 20 cm from the center of the mouth opening. Two other DPA 4060 microphones were placed at the top of the pinnae of each ear to simulate the microphone positions of BTE HAs.

For each of the 15 sentences, recordings were made with the listening HATS angled from 0 to $90°$ in $1°$ steps.
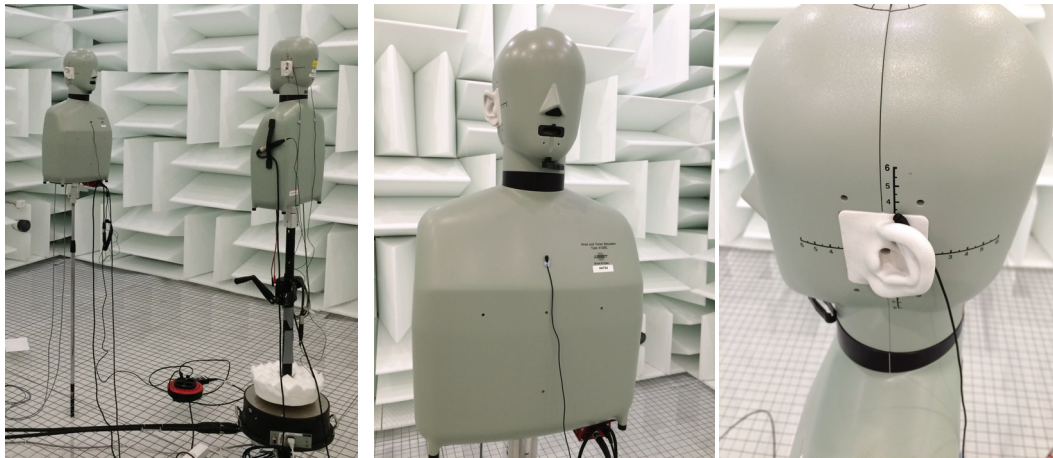
Fig. 1: Set-up for recording acoustic stimuli (left panel). Position of remote mic on "talking" HATS (middle panel). Position of microphones on "listening" HATS (right panel)

Participants

15 normal-hearing (NH) and 9 HI listeners participated in the study. All NH listeners had air-conduction audiometric thresholds below 25 dB between 125 Hz and 4 kHz. The average audiogram of the HI listeners is plotted in Fig. 2.

Procedure

A 3-interval 3-alternative forced-choice paradigm using a 1-up 2-down adaptive rule was used to estimate MAA. Between each run of the adaptive procedure, the target angle (i.e., the direction used in two of three intervals) was roved uniformly between 40–50°. At the start of each run, the initial angular difference was always as large as possible. The initial step size was 8° and the step size was halved after each reversal until the minimum step size of 1° was reached. Runs were terminated after four reversals and the threshold was estimated as the mean of the angular difference between target and presented angular value at the last two reversals. Sentences were randomly chosen between trials but remained the same within each 3-interval triplet.

For each presentation, two audio files were loaded, one with the HA signal and one with the RM signal. According to the current state of the test, a gain of either 0, −3 or −6 dB was applied to the RM signal and a delay of either 0, 10, 20, 40, 60, 80 or 100 ms was applied to the RM signal. After applying gain and delay to the signals, they were mixed into one stereo file, which was then presented to the listener via headphones.

A further control condition, in which only the HA recordings were presented, was conducted to estimate the baseline MAA performance with microphones in BTE HA positions.
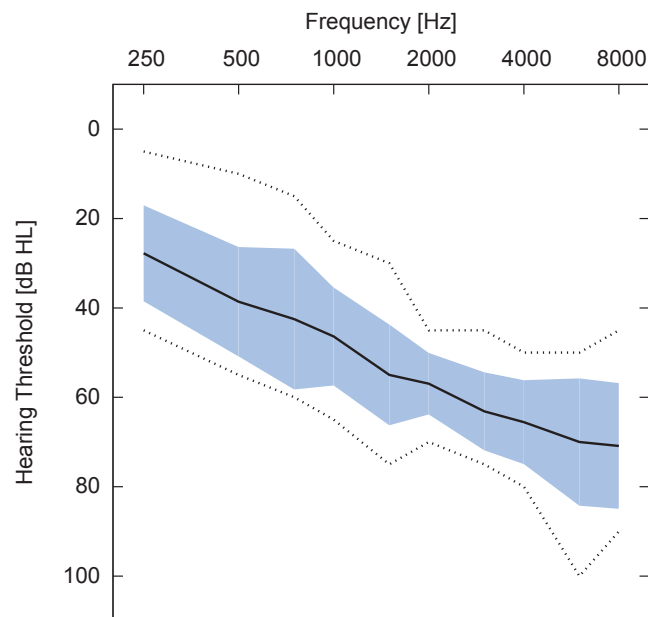
Fig. 2: Audiometric thresholds of the hearing-impaired listeners. The solid black line indicates the mean hearing threshold, the shaded region indicates one standard deviation, and the dotted lines indicate minimum and maximum measured thresholds.

To compensate for reduced audibility, the mixed stimuli was further amplified using the linear Cambridge Formula (Moore and Glasberg, 1998) for each individual HI listener.

No correction was applied to compensate for the acoustic delay between RM and HA microphones, which was approximately 2.5 ms. Thus, in the 0 ms condition, RM signals preceded HA signals.

RESULTS

The average MAA as a function of relative delay for the three relative gains is plotted in Fig 3. The lower dotted line in each panel indicates the MAA threshold for the control condition when no RM signal was mixed with the HA recordings. Thus, this estimates the minimum MAA listeners could achieve when using ideal BTE HAs. As expected, NH listeners exhibited lower MAA overall than the HI listeners.

For the normal hearing listeners, two overall trends were observed: (1) MAA thresholds decreased as the RM gain was decreased; (2) MAA thresholds were minimized when the relative delay was between 10–20 ms. The same trends were observed in the HI listeners but the size of the effects were smaller.
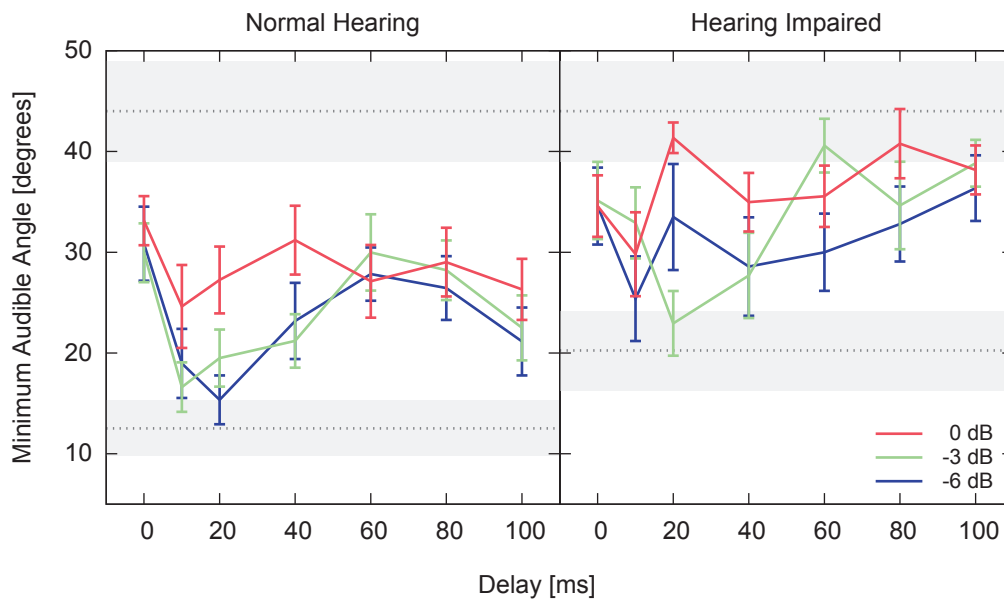
Fig. 3: Average MAA as a function of relative delay for three different relative gains of RM vs HA microphone signals for normal-hearing (left panel) and hearing-impaired listeners (right panel). The lower dotted line and shaded area indicate the average MAA for the HA only condition. The upper dotted lines indicate chance performance. The bars and shaded areas indicate standard error.

These observations were confirmed by the results from a repeated measures ANOVA with gain and delay as within-group and hearing status as between-group factors. Statistically significant main effects of gain [$F(2,46) = 9.002$, $p = 0.01$] and delay [$F(6,138) = 7.637$, $p < 0.001$] and hearing status [$F(1,23) = 304.827$, $p < 0.001$] were observed.

DISCUSSION

For the NH listeners, MAA was smallest when RM gain was reduced and the RM delay was approximately 10–20 ms. Indeed, for relative gains of -3 and -6 dB, NH listeners exhibited MAA thresholds that were similar to those they achieved with only the microphone signals from the simulated BTE positions. A similar pattern of results was obtained from the HI group of listeners. However, their overall MAA thresholds were higher and the size of the effects were smaller.

For speech, previous studies have found echo thresholds ranging from 30–50 ms (Lochner and Burger, 1958; Haas, 1951). Thus, a relative delay of the RM signal of 10–20 ms should result in a fused percept of a single talker and the precedence effect suppresses the incongruent spatial information presented by the RM signal.

Some of the largest MAA thresholds were obtained in the 0-ms delay conditions. However, no correction was applied to compensate for the acoustic delay between RM and HA microphone positions. As a result, in the 0-ms condition, RM signals preceded HA signals by approximately 2.5 ms. Thus, it is not surprising that larger MAA thresholds were observed in these conditions as the spatial cues available from the HA microphones were likely suppressed by the precedence effect.

Overall, the present study suggests that to achieve optimal spatial perception, the signal from the RM microphone should be mixed with both little gain as possible and a relative delay of 10–20 ms (after compensating for the difference in acoustic delay between the RM and HA). However, there are some potential issues with this advice.

First, the reason for employing an RM is to improve speech intelligibility and/or reduce listening effort by providing the HI listener with a higher quality speech signal (i.e., one with a higher SNR and/or direct-to-reverberant energy ratio). Thus, reducing the gain of the RM reduces its potential benefit for speech intelligibility and listening effort.

Second, in current devices, the delay caused by mixing the RM signal is heavily influenced by the communication protocols used in the digital transmission of the signal from RM to HA. As the actual delay of each device is proprietary, it is difficult to compare our results to the delays that are present in products that are currently available on the market. That said, common digital transmission protocols, such as Bluetooth, can result in delays of 100 ms or more. For use cases involving multiple RMs, much lower delays are likely desirable.

Assuming a sufficiently low-latency transmission protocol was employed, achieving the suggested delay target requires the HA to estimate and compensate for the acoustic delay differences between RM and HA microphones. While this presents a technical challenge, other research into improving selective attention in HI listeners assumes similar requirements (e.g., Favre-Felix *et al.*, 2017).

In the present study, the HA signals were simulated using omnidirectional microphones positioned at the pinnae. Thus, the signals presented replicated an "ideal" HA (i.e., full bandwidth) and did not include standard HA signal processing that might affect spatial perception (e.g., directional microphones/beamforming, compression, and noise-reduction). Further, the signals were recorded in anechoic conditions. Thus, MAA thresholds in more realistic conditions are likely to be larger.

CONCLUSION

Based on the results from the present study, the detrimental effects of an RM on localization can be minimized by targeting a relative delay of 10–20 ms between the RM and HA signals. Further, the gain of the RM should be reduced as much as is possible while still maintaining its beneficial effects on speech intelligibility.

ACKNOWLEDGEMENTS

REFERENCES

American Academy of Audiology Clinical Practice Guidelines (**2011**). "Remote microphone hearing assistance technologies for children and youth from birth to 21 years."

Boothroyd, A. (**2004**). Hearing aid accessories for adults: The remote FM microphone. Ear Hearing, **25**, 22-33. doi: 10.1097/01.AUD.0000111260.46595.EC

Favre-Felix, A., Graversen, C., Dau, T., and Lunner, T. (**2017**). "Steering of audio input in hearing aids by eye gaze through in-ear electrodes," in *Adaptive Processes in Hearing*, Proc. ISAAR, **6**, in press.

Haas, H. (**1951**). "On the influence of a single echo on the intelligibility of speech," Acustica, **1**, 48-58.

Hawkins, D.B. (**1984**). "Comparisons of speech recognition in noise by mildly-to-moderately hearing-impaired children using hearing aids and FM systems," J. Speech Hear. Disord., **49**, 409-418. doi: 10.1044/jshd.4904.409

Lochner, J.P.A., and Burger, J.F. (**1958**). "The subjective masking of short time delayed echoes by their primary sounds and their contributions to the intelligibility of speech," Acustica, **8**, 1-10.

Moore, B.C.J., and Glasberg, B.R. (**1998**). "Use of loudness model for hearing-aid fitting. I. Linear hearing aids," Brit. J. Aud., **32**, 317-335. doi: 10.3109/03005364000000083

Nábělek, A.K., and Donahue, A.M. (**1986**). "Comparison of amplification systems in an auditorium," J. Acoust. Soc. Am., **79**, 2078-2082. doi: 10.1121/1.393167

Ross, M., and Giolas, T.G. (**1971**). "Effect of three classroom listening conditions on speech intelligibility," Am. Ann. Deaf, **116**, 580-584.

Sørensen, A.J., Weisser, A., and MacDonald, E.N. (**2017**). "Preliminary investigation of the categorization of gaps and overlaps in turn-taking interactions: Effects of noise and hearing loss," in *Adaptive Processes in Hearing*, Proc. ISAAR, **6**, in press.