A bilateral hearing-aid algorithm that provides directional benefit while preserving situational awareness

TOBIAS PIECHOWIAK[1,*], ROB DE VRIES[2], CHANG MA[3], AND ANDREW DITTBERNER[3]

[1] *GN ReSound A/S, Ballerup, Denmark*

[2] *GN ReSound, Dept. R&D, Eindhoven, The Netherlands*

[3] *GN ReSound North America, Glenview, IL, USA*

A directional filter or beamformer is a classical approach to ensure speech intelligibility by suppressing distracting sounds from certain directions. However, they have some challenges on their own: (a) white-noise gain, (b) diminished benefit in reverberation, and (c) 'off-axis' audibility problems. In this study a new algorithm which is part of ReSound's Linx3D's Bilateral Directionality III is introduced. It is designed to provide good *situational awareness* (SA) while mainting *directional benefit* (DB). SA is maximized by combining the sensitivity of both left and right hearing aids to create a true binaural omnidirectional sensitivity pattern. DB is ensured by promoting the better-ear effect and thus allowing for better separation of sounds.

INTRODUCTION

The primary purpose of a hearing aid is to restore audibility of a target signal. The classical approach is to measure a pure-tone audiogram and apply frequency dependent gains on the microphone input signals. However, it often does not alleviate the problems hearing impaired have in noisy environments (Kochin, 2010). Directional filters (or beamforming filters) are one attempt to further address this problem by suppressing distracting sounds from certain (a-priori) known directions. They have some challenges on their own as, e.g.:

1. High white noise gain, i.e., creation of noise because of partially equalizing for inherent low-frequency roll-off;

2. Directional benefit dimishes quickly when reverberation is added (Ricketts, 2003); and

3. Directional filters impede 'off-axis' listening. Sounds from the side or rear are attenuated and might become inaudible creating problems when new sounds are introduced.

In this study we introduce a new (bilateral) algorithm, within ReSound's Linx3D Bilateral Directionality III, that is primarly targeting this last challenge, to promote

*Corresponding author: tpiechowiak@gnresound.com

Proceedings of the International Symposium on Auditory and Audiological Research (Proc. ISAAR), Vol. 6: Adaptive Processes in Hearing, August 2017, Nyborg, Denmark. Edited by S. Santurette, T. Dau, J. C.-Dalsgaard, L. Tranebjærg, T. Andersen, and T. Poulsen. The Danavox Jubilee Foundation, 2017.
 ISBN: 978-87-990013-6-1.

better ‘off-axis’ listening while simultaneously not giving up on the directional benefit a beamformer provides for, e.g., increasing speech intelligbility. With other words, the new algorithm tries to combine the two complementary concepts *situational awareness* (SA) and *directional benefit* (DB). Basically, the new algorithm in itself constitutes a beamformer whose target is to provide a combined quasi-omnidirectional response across ears while simultaneously maintaining a large head-shadow (better-ear effect) that helps with source separation and speech intelligibility (Bronkhorst, 1988).

THE ALGORITHM

The algorithm processing scheme is depicted in Fig. 1.

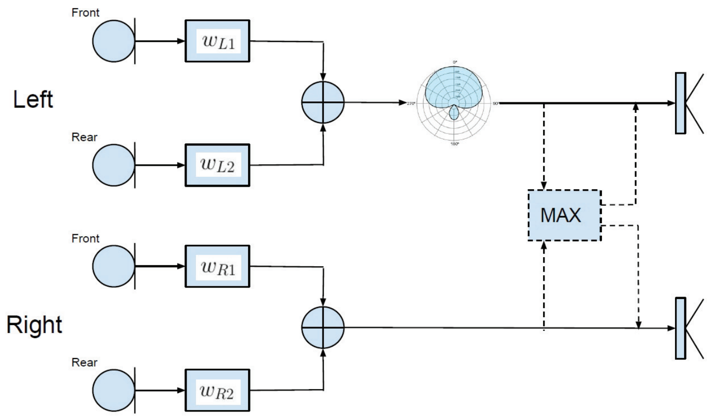

Fig. 1: Flowchart of the algorithm. The weights w_{Ri} are optimized in the way to give a quasi-omnidirectional response across ears. The weights on the left side w_{Li} are fixed to generate a hypercardioid at the left ear. Weights are optimized offline so the algorithm shown is not adaptive.

The $w_i(n)$ are fixed finite-impulse-response (FIR) filters associated with the i-th mircophone at the left ($w_{Li}(n)$) and right side ($w_{Ri}(n)$). The output of the right, respectively left side can be defined as

$$L(n,\Theta) = \sum_{i=1}^{2} w_{Li}(n) * h_{Li}(n,\Theta) * s(n) \qquad \text{(Eq. 1)}$$

$$R(n,\Theta) = \sum_{i=1}^{2} w_{Ri}(n) * h_{Ri}(n,\Theta) * s(n) \qquad \text{(Eq. 2)}$$

where $s(n)$ is the acoustic source and $h_i(n,\Theta)$ the hearing aid related impulse responses associated with the i-th microphone and azimuth angle Θ. Note, that the $w_{Li}(n)$ are not part of the optimization scheme. They are used to generate a hypercardiod on the left side as depcited in the figure and are fixed during optimization. The hypercardioid is part of providing directional benefit.

The algorithm itself is achieved by optimizing the weights $w_{Ri}(n)$ in a least-square sense:

$$\underset{w_{Ri}}{\operatorname{argmin}}\{Var\left[max(L(n,\Theta_k),R(n,\Theta_k))\right]\} \quad k \in \{1...K\} \qquad \text{(Eq. 3)}$$

where K is the number of sampled azimuth angles. In this study a resolution of 10° was used for that purpose. Minimizing the variance (Var) results in a quasi-omnidirectional response across ears aiming to provide high SA. Note, that the optimization is performed off-line, so no information is exchanged between devices during use of the algorithm. The resulting intensity plots for the left, respectively right side can be seen in Fig. 2.

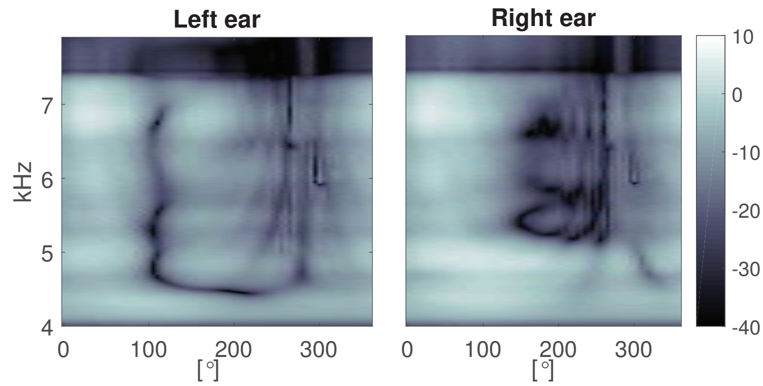

Fig. 2: Intensity plots for the optimized beampattern of the left and right side across azimuth and frequencies. Bright color illustrates high intensity. Responses are normalized to the maximum of the front microphone.

THE METRIC

The algorithm poses a fundamental question: What would be an appropriate metric for characaterizing the two-fold purpose of the algorithm? For example, the *Directivity Index* (DI) has usually been used to characterize the strength of the directional benefit of a beamformer that can be calculated on a manikin (Dittberner, 2007). DI in general correlates quite well with directional benefit perceived by subjects (Dittberner, 2007). However, this metric would not be able to account for SA since SA and DB are complementary concepts. Thus, the need for a reliable metric arises that considers both SA and DB simultaneously. For this purpose, a new metric is introduced. It consists of two indices, (a) the Situational Awareness Index (SAI) and (b) the Better-Ear Index (BEI).

Formally, SAI and BEI are defined as

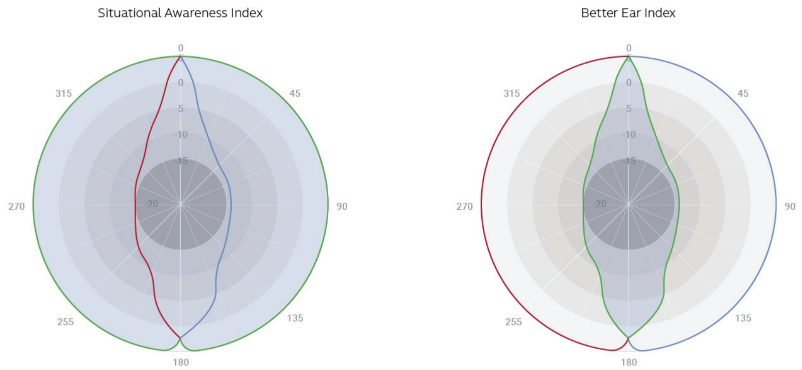

Fig. 3: Calculation of the metric. Left panel: Maximum power of across ears illustrates Situational Awareness Index (SAI). Right panel: Minimum power across ears (green line) illustrates Better-Ear Index. Grey shaded areas illustrate meaning of the indices.

$$SAI = 10 \cdot log_{10}\left(\frac{Std(max(P_L(\Theta_k),P_R(\Theta_k))}{\overline{max(P_L(\Theta_k),P_R(\Theta_k)}}\right) \qquad \text{(Eq. 4)}$$

$$BEI = 10 \cdot log_{10}\left(\frac{min(P_L(\Theta_k),P_R(\Theta_k))}{\overline{min(P_L(\Theta_k),P_R(\Theta_k)}}\right) \quad k \in \{1...K\} \qquad \text{(Eq. 5)}$$

where P_L, respecticely P_R are the powers of the left, respectively right side at angles Θ_k, Std denotes standard deviation and $\overline{...}$ the mean. Note, that $P_L \propto \sum L(n,\Theta)^2$ and $P_R \propto \sum R(n,\Theta)^2$. The metric is defined as

$$Metric = BEI - SAI \qquad \text{(Eq. 6)}$$

In Fig. 3 the meaning of these indices is illustrated by the grey shaded areas. Note, that BEI resembles the definition of the classical directionality index (DI). However, while the perceptual effect of DI in its classical definition is based on the attenuation of sound from other than the frontal direction, providing directional benefit is achieved by the better-ear effect that. It describes a strategy of listening to a sound primarily through the ear at which the sound is strongest. It leads to a better separation of sources from different directions which helps with speech intelligibility. Additionally, the better-ear effect facilitates loudness summation by boosting frontal signals by 3 dB in contrast to sound coming from arbitrary other directions. Equation 6 now makes it possible to determine metric values for different algorithms characterizing how well they provide SA while at the same time maintaining DB. Figure 4 gives an example for the metric for an omnidirectional microphone on both ears and the new algorithm measured on a KEMAR manikin.

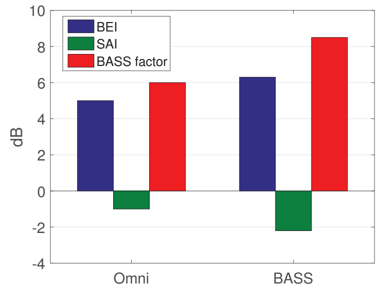

Fig. 4: SAI, BEI and the metric for an symmetric omnidirectional response on both ears and the new algorithm.

CORRELATION BETWEEN METRIC AND PERCEPTION

One question that arises from the previous sections is "what is the correlation between the metric and perception?" This section will try to address this question by measuring situational awareness and directional benefit for different metric values. In order to generate realistic sound environments and control the value of the metric, a combination of a virtual test environment with a room simulation software was applied in this study.

Room simulation software (MCRoomSim)

The room simulation software that was used in this study is MCRoomSim, a free-ware software tool (Wabnitz et al., 2010). MCRoomSim simulates both specular ('ray-tracing') and diffuse reflections in a rectangular 'shoebox' environment. It provides a MATLAB interface that allows for high level programming and set-up of simulation parameters. The output of the simulation is a matrix of room-impulse-responses from each source to each receiver channel which can be directly used in any audio application as was done in a speech-on-speech intelligibility test for this study.

Test setup

Figure 5 illustrates the setup for the measurement of situational awareness (a + b) and directional benefit (c):

For SA two distracting speech streams (red symbols) are presented from the frontal hemisphere while the target speech (green symbol) is presented either from the left or right side ('off-axis'). Intelligibility is measured for both situations independently and the obtained thresholds are averaged. In terms of our abovementioned example this would correspond to measure how sensitive one is to detecting sounds around you.

Values of metric and test environment

The polar patterns yielding different values of the metric which were applied in this study are shown in Fig. 6 . These patterns will be applied in MCRoomSim and were

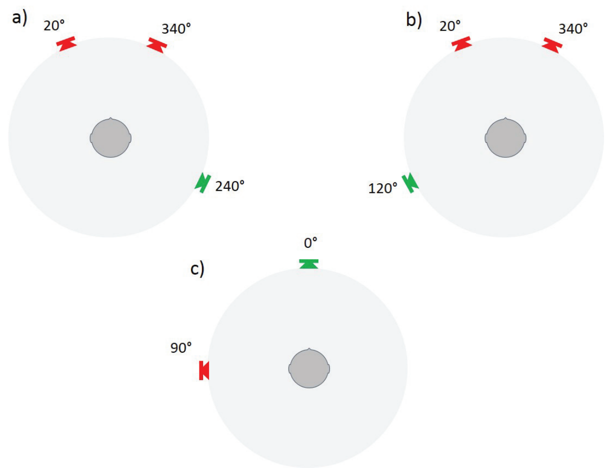

Fig. 5: Setup for measuring situational awareness (a + b) and directional benefit (c). Dark symbols denote distracting sound streams, light symbols target sound.

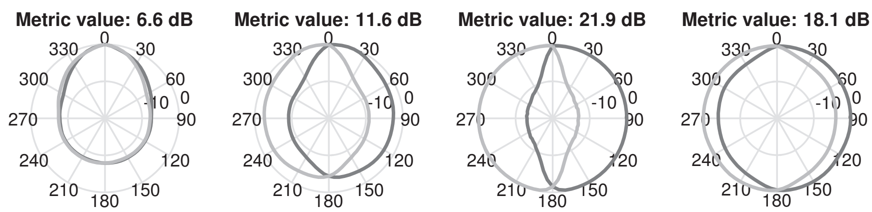

Fig. 6: Polar plots corresponding to increasing the values of the metric from left to right panel. Low values are characterized by a low situational awareness as well as small head shadow. Colors indicate sensitivity for the right (dark) and left ear (light).

applied as a direction-dependent hearing aid receiver gain. The simulated room had a low reverberation with an average broadband reverberation time of around 0.2 s. The speech reception thresholds were obtained with the help of a Danish HINT (Nielsen, 2014) implemented in MATLAB.

Subjects

Twelve normal-hearing subjects participated in the test.

RESULTS

Mean results for situational awareness and speech intelligibility are seen in the left and respectively right panel of Fig. 7. Lower speech reception thresholds (SRT) indicate a better performance. Dashed grey lines indicate trend lines. For situational awareness, the best linear fit to the data is given by $-0.53\frac{dB}{dB} \cdot x + 8.02\,dB$ while for directional benefit the best fit is $-0.1\frac{dB}{dB} \cdot x - 8.34\,dB$. Threshold values for directional benefit are lower than for situational awareness. This is likely due to two effects: Thresholds for spatial separated sounds are lower for a single masker than for multiple masker sounds and the target receives a 3-dB boost due to addition of sounds from the frontal directions for all the investigated metric values.

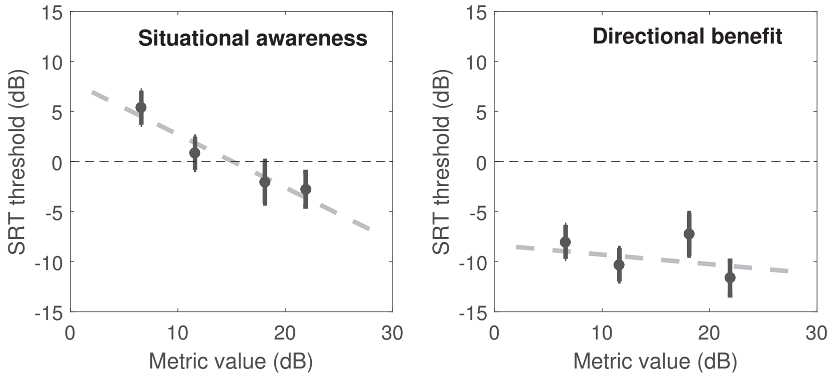

Fig. 7: Mean speech reception thresholds (SRT) for situational awareness and directional benefit. Dashed grey lines indicate trend lines.

The increase in directional benefit with larger metric values is not as large as in the case of situational awareness. However, there is one reason that the definition of the new metric makes sense: For this paradigm a higher metric value would result in less ‘off-axis’ attenuation and higher SRTs for a target from the front simply because less masking energy is attenuated. However, looking at the data this is clearly not the case. It indicates that the increase in masker energy with increasing metric values can indeed be compensated through the use of a larger head shadow and thus spatial separation that follows from an increasing BEI and thus higher metric values.

CONCLUSIONS

The new algorithm in Linx3D’s Bilateral Directionality III tries to combine two aspects in a single hearing-aid microphone mode:

1. Situational awareness or ability to listen ‘off-axis’
2. Speech intelligibility, the ability to understand speech coming from a certain direction in most cases the frontal direction

In order to be able to quantify such an approach an effective metric needed to be developed. The new defined metric fulfills this ne ed. The main purpose of this study was to correlate the (objective) metric values with perceptual data. In general, the most important finding in this study is that the new metric seems to be an appropriate tool for collectively quantifying SA and DB for hearing aid algorithms.

REFERENCES

Bronkhorst, A.W., and Plomp, R. (**1988**). "The effect of head-induced interaural time and level differences on speech intelligibility in noise," J. Acoust. Soc. Am., **83**, 1508-1516.

Dittberner, A.B., and Bentler, R.A. (**2007**). "Predictive measures of directional benefit part 1: Estimating the directivity index on a manikin," Ear Hearing, **28**, 26-45.

Kochkin, S. (**2010**). "Why my hearing aid is in the drawer: The consumer's perspective," Hearing Journal.

Nielsen, J.B., and Dau, T. (**2014**). "A Danish open set speech corpus for competing speech studies," J. Acoust. Soc. Am., **135**, 407-420.

Ricketts, T.A., and Hornsby, B.W.Y.(**2003**). "Distance and reverberation effects on directional benefit," Ear Hearing, **24**, 472-484.

Wabnitz, A., Epain, N., Craig, J., and van Schaik, A. (**2010**). "Room acoustic simulation for multichannel microphone arrays," Proc. Int. Symp. Room Acoustics, 1-6.