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ORIGINAL ARTICLE

A Comparison of Frequency-invariant Beamforming Algorithms for Hearing Aids: Differential Microphone-based Beamformers and the Broadband Beamformer

Kyeongwon Cho, Kyoung Won Nam, Jun Chang Lee, Sung Hwa Hong, See Youn Kwon, Jonghee Han, Dongwook Kim, Sangmin Lee and In Young Kim

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Abstract

Purpose Recently, some research groups have suggested the possibility of using the broadband beamformer (BBF) algorithm for hearing aid applications. However, there have been no previous reports to have quantitatively evaluated the relative performance between conventional differential microphone (DM)-based frequency-invariant beamforming algorithms and the broadband beamformer.

Methods In this study, we evaluated the performance of DM-based beamformer algorithms and the BBF algorithm *in vitro* using the four objective indices of signal-to-noise ratio (SNR), perceptual evaluation of speech quality (PESQ), noise distortion (C_{bak}) and weighted spectral slope (WSS) in a non-reverberant environment.

Results The experimental results showed that the DM-based algorithms were superior in terms of SNR, WSS and C_{bak}, and the BBF algorithm was superior in terms of PESQ.

Conclusions Considering the limited performance of hearing aid processors and the experimental results, DM-based frequency-invariant algorithms with a first-order compensation filter are more feasible for real hearing aids. However, additional *in vitro* and clinical evaluations are required to more accurately verify the clinical feasibility of these algorithms.

Keywords Digital hearing aids, Beamforming, Directional microphone

Kyeongwon Cho[†], Kyoung Won Nam[†], Jun Chang Lee, In Young Kim (✉)
Department of Biomedical Engineering, Hanyang University, Seoul, Korea
Tel : +82-2-2291-1713 / Fax : +82-2-2220-4949
E-mail : iykim@hanyang.ac.kr

Sung Hwa Hong
Department of Otorhinolaryngology-Head and Neck Surgery, Sungkyunkwan University, Seoul, Korea

See Youn Kwon, Jonghee Han, Dongwook Kim
Samsung Electronics, Yongin, Korea

Sangmin Lee
Department of Electronic Engineering, Inha University, Incheon, Korea

[†]Kyeongwon Cho and Kyoung Won Nam equally contributed to this paper and should therefore be regarded as equivalent authors.

INTRODUCTION

The speech recognition ability of sensorineural hearing-impaired individuals decreases in noisy environments because of the deteriorated time and spectral resolution of auditory systems [1, 2]. To improve speech intelligibility in noisy situations for hearing-impaired people, most hearing aid systems utilize various noise reduction algorithms [3] such as beamforming technology, which considers all sounds from directions other than the direction of a speech counterpart to be environmental noise and forcibly attenuates the noise components [4, 5]. Most general algorithms for this beamforming purpose in the hearing aid field are first-order differential microphone (DM1) algorithms [6] and two-stage differential microphone (DM2) algorithms [7], but these base algorithms have an intrinsic problem in the output signal attenuation in the low-frequency range, which can distort the original speech signal. To solve this problem of attenuation in the low-frequency range in real hearing aid systems, various low-frequency compensators such as low-pass filters (LPFs) and integrators are added to the DM1 or DM2 algorithms [6, 8] to implement a frequency-invariant beamformer that can minimize the output attenuation in the low-frequency range.

The broadband beamformer (BBF) algorithm, which represents a frequency-invariant beam pattern regardless of the frequency variation, has been utilized in various speech-signal processing systems comprised of multiple-microphone arrays, and several research groups [9, 10] recently suggested the possibility of applying the BBF algorithm to hearing aid systems that contain two microphones with relatively short interspacing as a frequency-invariant beamformer. However, there have been no previous reports to have quantitatively evaluated the relative performances between DM1- or DM2-based beamforming algorithms or the BBF algorithm in real hearing aid environments.

In this study, we evaluated the relative performances of both of the DM-based frequency-invariant beamforming algorithms and the BBF algorithm, which were modified to satisfy the geometric requirements of the mini behind-the-ear (BTE) type hearing aid *in vitro*. All the algorithms were implemented using MATLAB Simulink (Version 7.12.0.635; Mathworks Inc., MA, USA), and the evaluation was performed in a non-reverberant environment. Six independent infinite-impulse-response (IIR)-type low-pass filters were added to the base DM1 or DM2 algorithms to implement the frequency-invariant characteristic, and four common objective indices were calculated to compare the performances of each of the algorithms.

MATERIALS AND METHODS

Frequency-invariant beamforming algorithms

DM-based beamforming algorithms

The DM1 algorithm was implemented as described previously by of Elko *et al*. [6]. The input signal of a rear microphone was time-delayed at a constant α ($0 < \alpha <$ external delay) by internal software, and this time-delayed input signal was subtracted from the input signal of a front microphone to obtain a cardioid-shaped output polar pattern. The null direction of the output polar pattern was controlled in the range of 90 to 180° by adjusting the value of the constant α. The DM2 algorithm was implemented in accordance with the method described by Luo *et al*. [7]. Both the front- and rear-cardioid patterns were calculated and the rear-cardioid pattern was multiplied by a constant β ($0 < \beta < 1$) and subtracted from the front-cardioid pattern to obtain an output polar pattern. The null direction of the output polar pattern was controlled in the same range by adjusting the value of the constant β.

To achieve the frequency-invariant characteristic by compensating for the output attenuation in the low-frequency ranges, each of six independent low-pass filters – first-, second- and third-order IIR Butterworth filters and first-, second- and third-order IIR Chebyshev type II filters – was added to the DM1 or DM2 algorithm. Since the low-frequency attenuation of the DM1 and DM2 algorithms was linear in an octave scale, the coefficients of each low-pass filter were calculated by the computer simulation to cancel out the low-frequency attenuation of the DM1 and DM2 algorithms. The filter specifications of each low-pass filter were set as listed in Table 1 in accordance with the request of the MATLAB Simulink. Fig. 1 illustrates the magnitude characteristics of the selected low-pass filters. The sampling frequency of the input signal was set to 16 kHz and the horizontal interspacing distance between the two microphones was set to 8 mm in accordance with the geometric requirements of the mini-BTE-type hearing aid housing used.

BBF algorithm

The BBF algorithm was implemented in accordance with the report of Mabande *et al*. [10]. In order to modify the original algorithm to fit the geometric specifications of the mini-BTE-type hearing aid, the number of microphones was reduced from three to two and the horizontal interspacing distance between the two microphones was set to 8 mm. The speed of sound in air was set to 343 m/sec for the calculations, and the length of the finite-impulse-response (FIR) filter was set to 512.

In vitro performance evaluations of the trial algorithms

Before the real-life recording experiments were carried out, two English sentence wave files and six environmental noise wave files – five for babble noise and one for white noise – were randomly selected from the IEEE sentence database [11] and the Phonak noise database [12], respectively. Sound recording was performed in the non-reverberant room (Length × Width × Height = 2720 × 2720 × 2100 mm) at the Samsung Medical Center. A KEMAR mannequin (Type 45BA; G.R.A.S. Sound & Vibration, Holte, Denmark) equipped with an artificial ear (Type KB0060; G.R.A.S. Sound & Vibration) was placed in the center of the room and a mini-

Table 1. Filter specifications used for filter coefficient determination.

Filters	Butterworth		Chebyshev	
	Cut-off frequency [Hz]	Pass-band Attenuation [dB]	Stop-band frequency [Hz]	Stop-band Attenuation [dB]
First order	300	3	4096	18
Second order	2000	3	7900	80
Third order	2000	3	4096	18

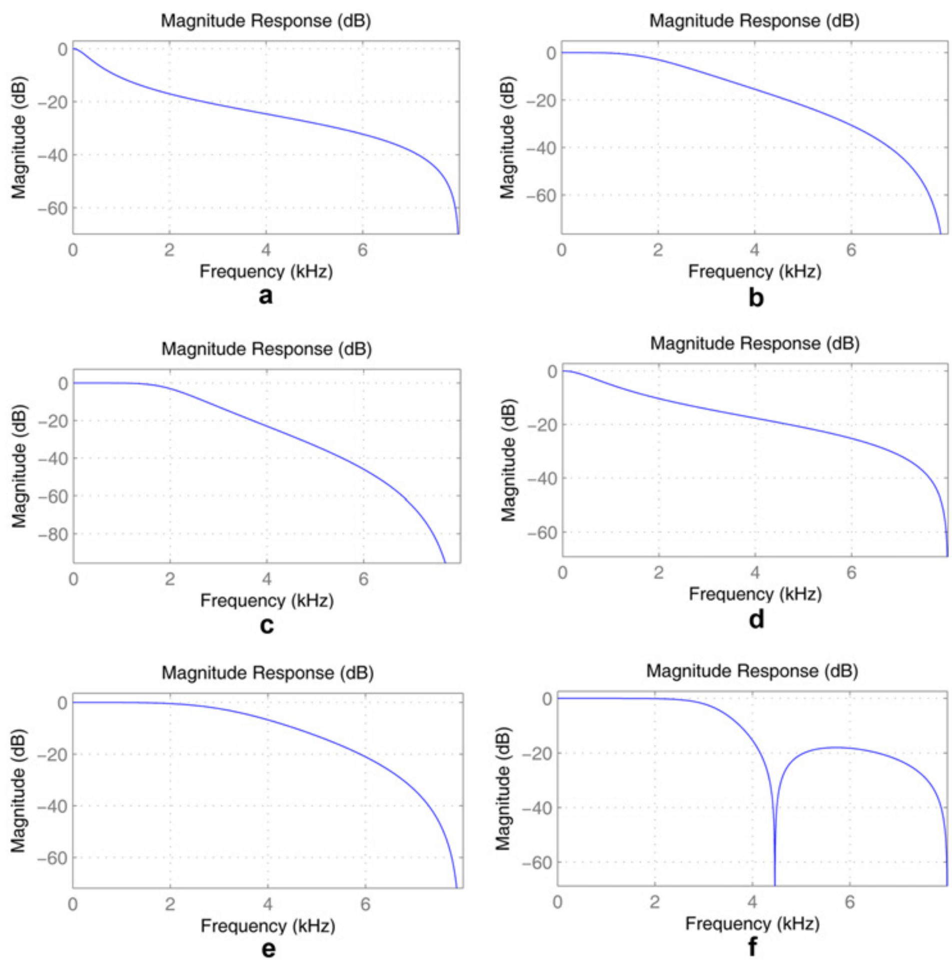

Fig. 1. Magnitude characteristics of the selected low-pass filters used to compensate for attenuation at low-frequency ranges in the DM-based algorithms. (a) 1st order Butterworth LPF (b) 1st order Chebyshev LPF (c) 2nd order Butterworth LPF (d) 2nd order Chebyshev LPF (e) 3rd order Butterworth LPF (f) 3rd order Chebyshev LPF.

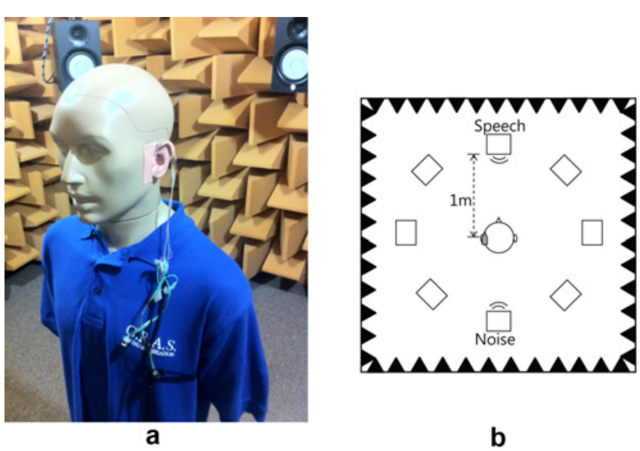

Fig. 2. Setting for sound recording. (a) A hearing aid housing was mounted on an artificial ear. (b) Eight speakers were arranged at equal distances from the KEMAR mannequin.

BTE-type hearing aid housing (DT3060; GN ReSound A/S, Ballerup, Denmark) was mounted on the left artificial ear (Fig. 2a). Eight loudspeakers (HS50M; Yamaha Corp., Hamamatsu, Japan) were arranged around the mannequin every 45° at a distance of 100 cm (Fig. 2b), and the heights of both the hearing aid housing and the speakers were fixed at 135 cm from the floor of the room. The input ports of the speakers were connected to the stereo-out port of a recording computer via an external sound card (Carat-RUBY, Styleaudio Corp., AZ, USA) and a router (ne24.24M; Ashly Audio Inc., NY, USA) so that the examiner could select any two speakers at the same time. In this study, two speakers at the 0° and 180° positions were utilized; the front speaker was used for the speech signal and the rear speaker for the noise signal to simulate a speech-in-noise situation. Two microphones in the hearing aid housing (EM-26392-C36; Knowles Electronics Inc., IL, USA) were connected to the recording computer via a MIDI device (Fast Track Ultra; Avid Technology Inc., MA, USA) and all input sound data were sampled at 16 kHz with 24-bit resolution. During the sound recording, the input signal-to-noise ratios (SNR) of the speech and noise signals were adjusted to -5, 0 and +5 dB and the same recording protocols were repeated in accordance with the variations of the three input SNR conditions, the two sentences, and the six environmental noises. This resulted in a total of 15 trial algorithms: DM1 only, DM1 plus first-, second- or third-order Butterworth filters, DM1 plus first-, second- or third-order Chebyshev type II filters, DM2 only, DM2 plus first-, second- or third-order Butterworth filters, DM2 plus first-, second- or third-order Chebyshev type II filters and BBF. After recording the sound, we quantitatively evaluated the performance of the 15 trial algorithms by comparing the values of four common objective indices were calculated for all of the input condition variations: the SNR improvement [5], perceptual evaluation of speech quality (PESQ) [13], weighted spectral slope (WSS) [14], and composite index for noise distortion (C_{bak}) [15].

RESULTS

Fig. 3 depicts the results of the frequency response and polar

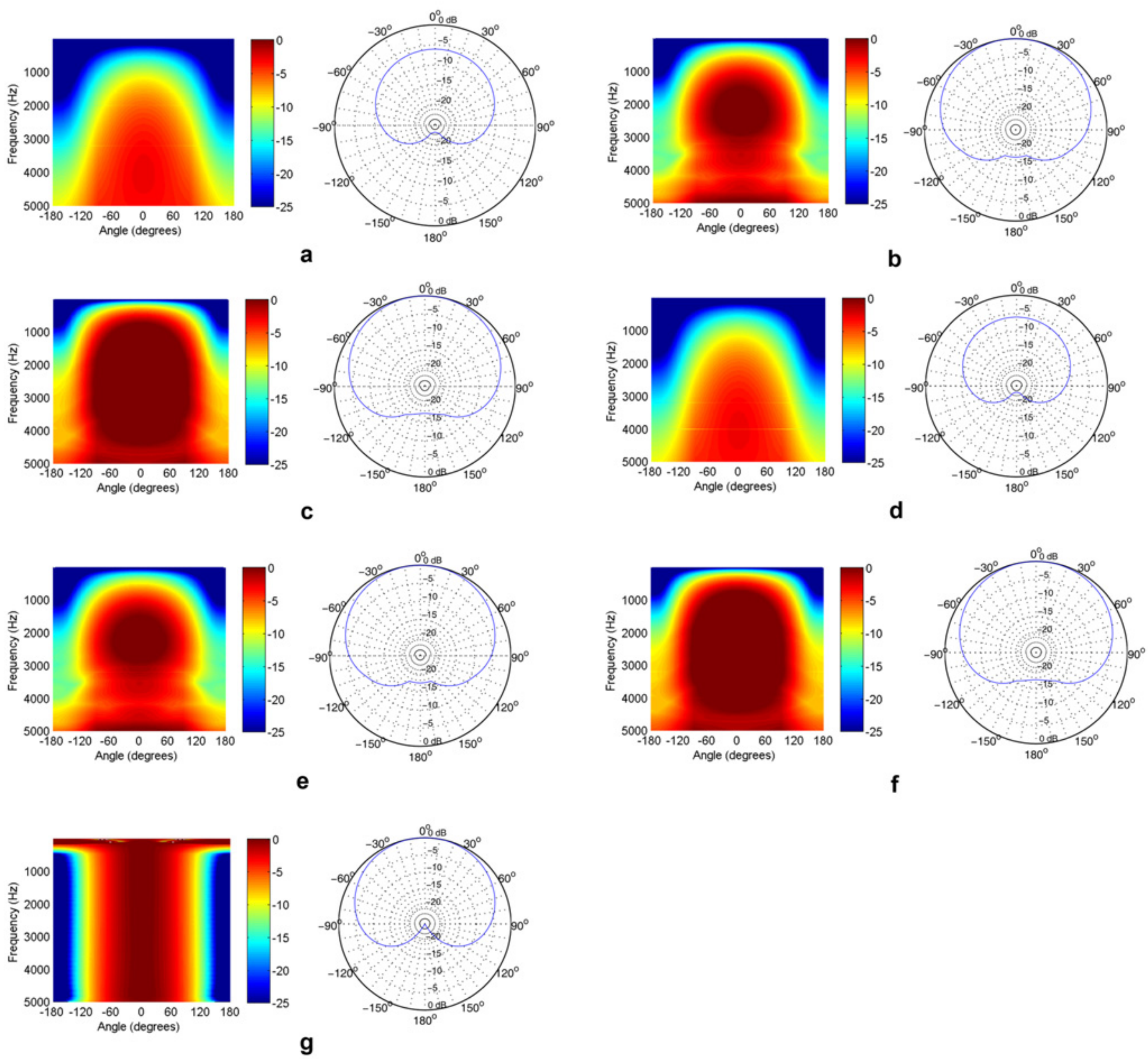

Fig. 3. Frequency responses in the 0 – 5000 Hz range and polar patterns of a 2-kHz pure-tone sine wave input for various trial algorithms: (a) DM1 only, (b) DM1 plus first-order Chebyshev LPF, (c) DM1 plus second-order Chebyshev LPF, (d) DM2 only, (e) DM2 plus first-order Chebyshev LPF, (f) DM2 plus second-order Chebyshev LPF and (g) broadband beamformer.

pattern simulations with the assumptions that there was no head-shadowing effect or sound reverberation, there was no sound loss during propagation, and microphones and sound sources were placed on the same surface. The beam patterns were drawn in the frequency range of 0 to 5000 Hz and the polar patterns were drawn from an input signal of a 2-kHz pure-tone sine wave.

Fig. 4 shows the improvement in output SNR in speech-in-babble noise situations. The BBF algorithm showed relatively superior performance compared with the DM1-only and DM2-only algorithms in most situations, but when the relevant IIR filter was added to those algorithms, the DM1- and DM2-based frequency-invariant beamformers performed relatively better than the BBF algorithm. For example, when the input SNR of the speech sample 1 was 0 dB, the SNR improvement was 3.40 ± 1.22 (mean ± standard deviation) and 3.40 ± 1.22 for the DM1-only and DM2-only, 3.91 ± 1.15 for the BBF, and 6.43 ± 1.40 and 6.43 ± 1.40 for the DM1 with first-order Butterworth filter and DM2 with first-order Butterworth filter. The degree of SNR improvement differed based on the filter type, filter order, kind of testing sentences, and input SNR conditions, and no regular trend was observed.

Fig. 5 depicts the PESQ indices for speech-in-babble noise situations. The PESQ values of the BBF algorithm were slightly higher than those of the DM1- and DM2-based algorithms in most situations; more specifically, when the input SNR of the speech sample 1 was 0 dB, the PESQ values were 2.24 ± 0.10 and 2.24 ± 0.10 for the DM1-only and DM2-only, 2.62 ± 0.06 for the BBF, and 2.46 ± 0.05 and 2.46 ± 0.05 for the DM1 with first-order Butterworth filter and DM2 with first-order Butterworth filter. In this case, the

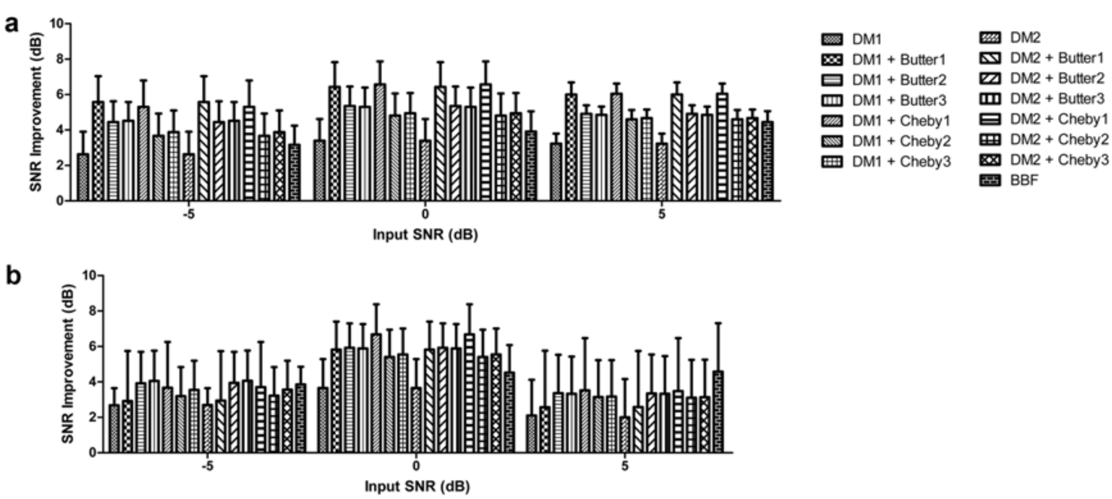

Fig. 4. The improvements in the output SNR for (a) speech sample 1 and (b) speech sample 2 in speech-in-babble noise situations.

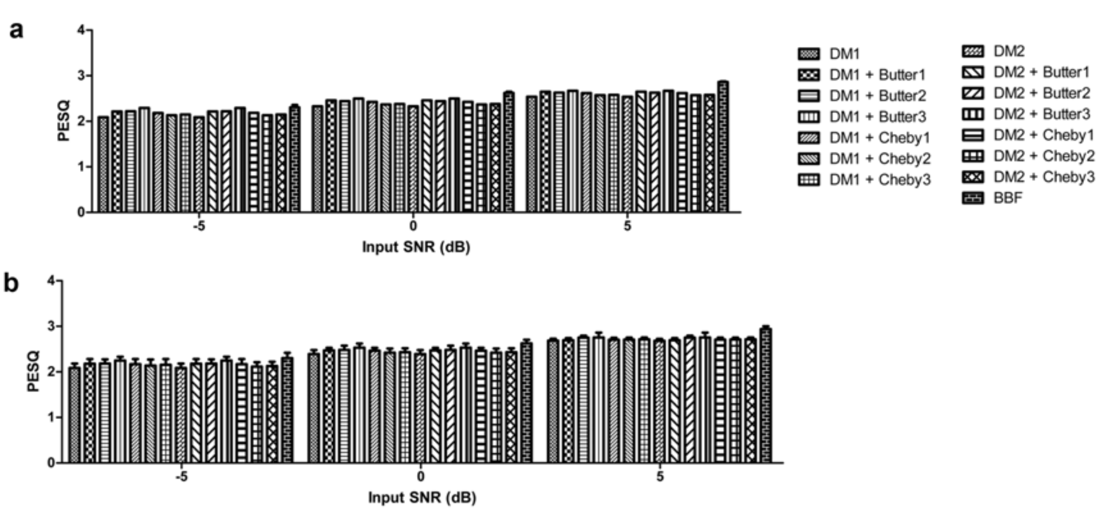

Fig. 5. The PESQ index measurements for (a) speech sample 1 and (b) speech sample 2 in speech-in-babble noise situations.

additional filters slightly improved the PESQ measurements but the degree of improvement was not significant.

Fig. 6 shows the WSS indices for speech-in-babble noise situations. The WSS values of the BBF algorithm were higher than those of the DM1- and DM2-based algorithms in most situations, which indicate that the performance of the DM1- and DM2-based frequency-invariant algorithms was superior to the BBF algorithm from the perspective of signal distortion. For example, when the input SNR of the speech sample 1 was 0 dB, the WSS values were 31.97 ± 0.89 and 31.97 ± 0.89 for the DM1-only and DM2-only, 43.85 ± 1.51 for the BBF, and 32.14 ±1.13 and 32.14 ± 1.13 for the DM1 with first-order Butterworth filter and DM2 with first-order Butterworth filter. As with the PESQ measurements, the additional filters did not have a significant effect on the WSS measurements.

Fig. 7 illustrates the C_{bak} index measurements for speech-in-babble noise situations. The C_{bak} values of the DM1- and DM2-only algorithms and the BBF algorithm were similar in most situations, and the additional filters slightly improved the C_{bak} measurements compared to the BBF algorithm, but the degree of improvement was not significant. For example, when the input SNR of the speech sample 1 was 0 dB, the C_{bak} values were 2.44 ± 0.08 and 2.44 ± 0.08 for the DM1-only and DM2-only, 2.36 ± 0.07 for the BBF, and 2.60 ± 0.04 and 2.60 ± 0.04 for the DM1 with first-order Butterworth filter and DM2 with first-order Butterworth filter.

Figs. 8, 9, 10 and 11 show the results of the output SNR improvement, PESQ, WSS and C_{bak}, respectively, for speech-in-white noise situations. The values of the indices were

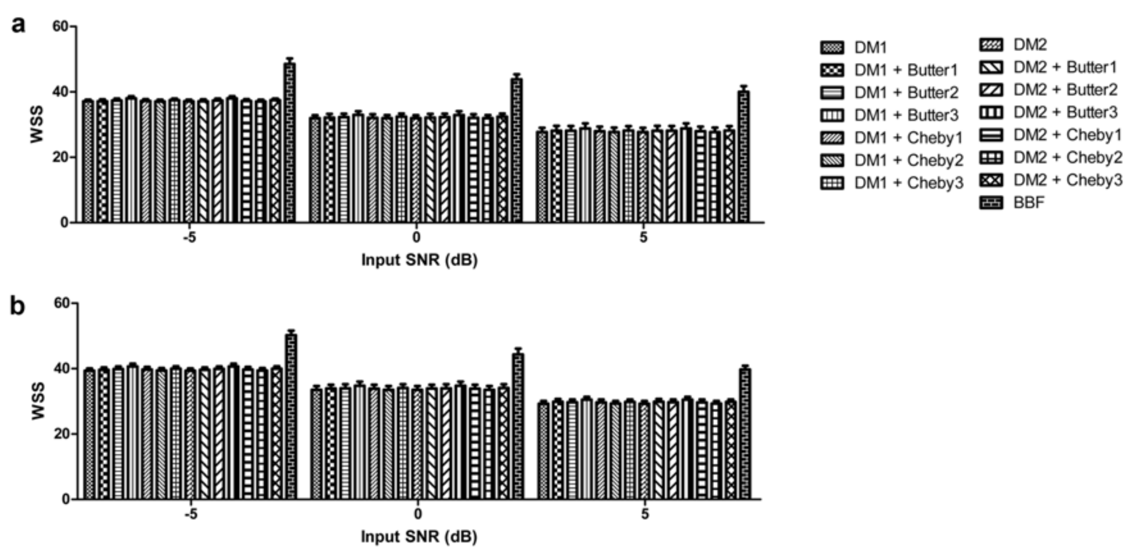

Fig. 6. The WSS index measurements for (a) speech sample 1 and (b) speech sample 2 in speech-in-babble noise situations.

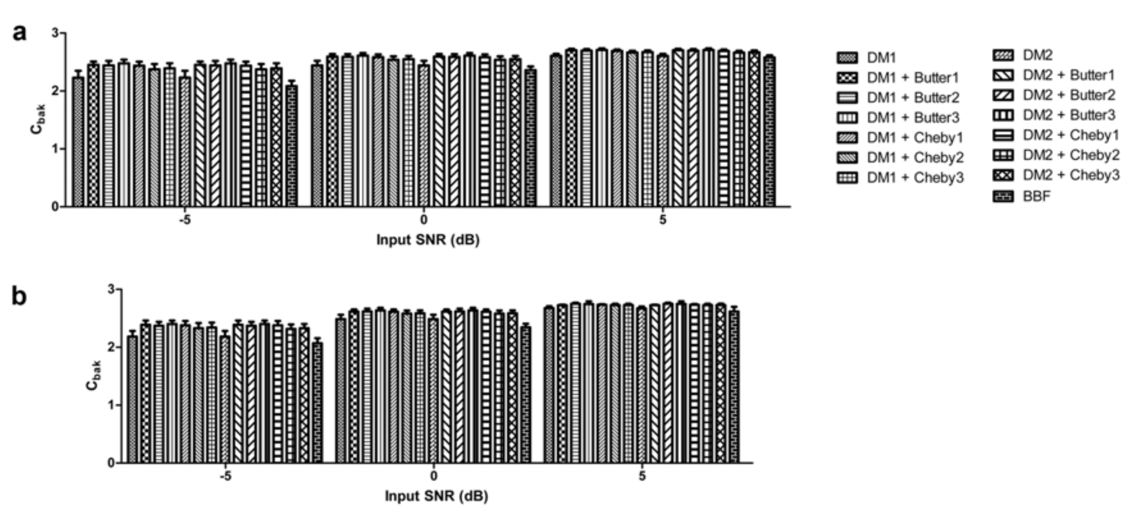

Fig. 7. The C_{bak} index measurements for (a) speech sample 1 and (b) speech sample 2 in speech-in-babble noise situations.

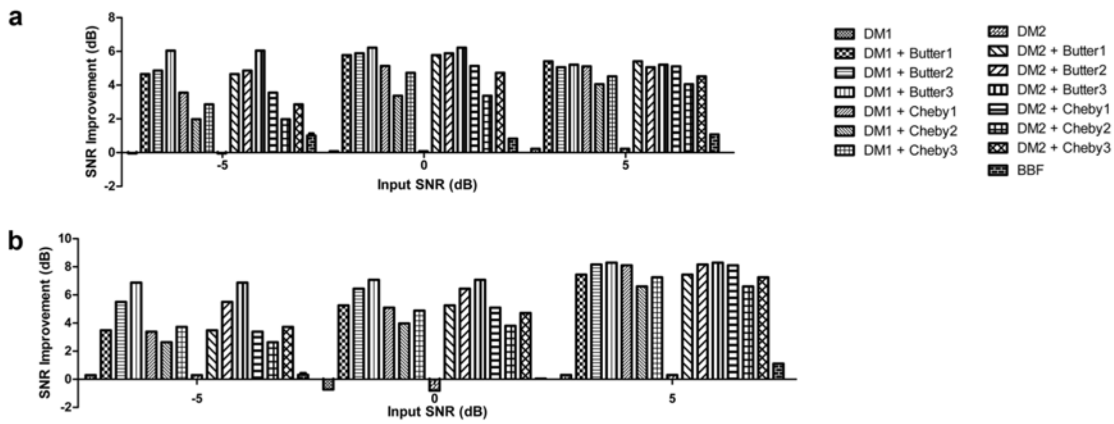

Fig. 8. The output SNR improvements for (a) speech sample 1 and (b) speech sample 2 in speech-in-white noise situations.

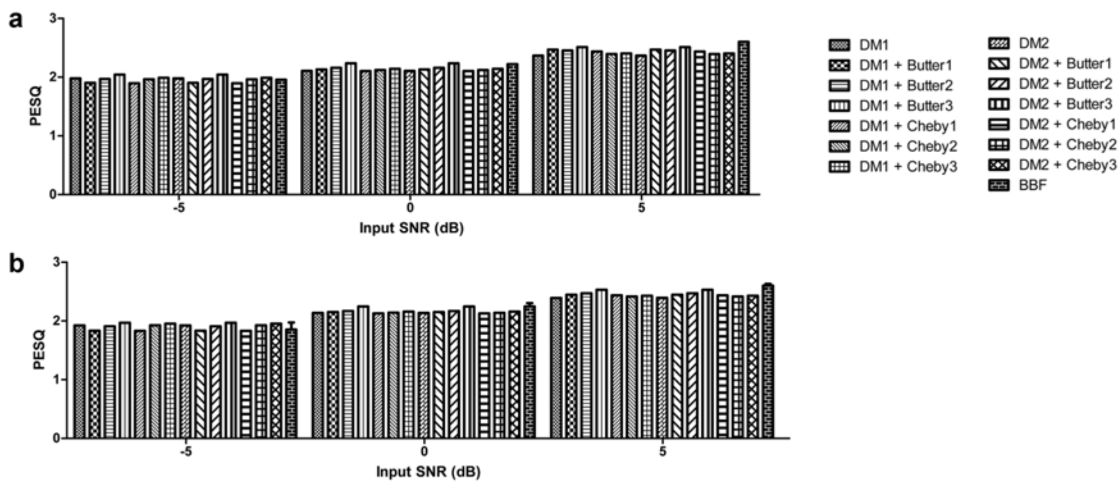

Fig. 9. The PESQ index measurements for (a) speech sample 1 and (b) speech sample 2 in speech-in-white noise situations.

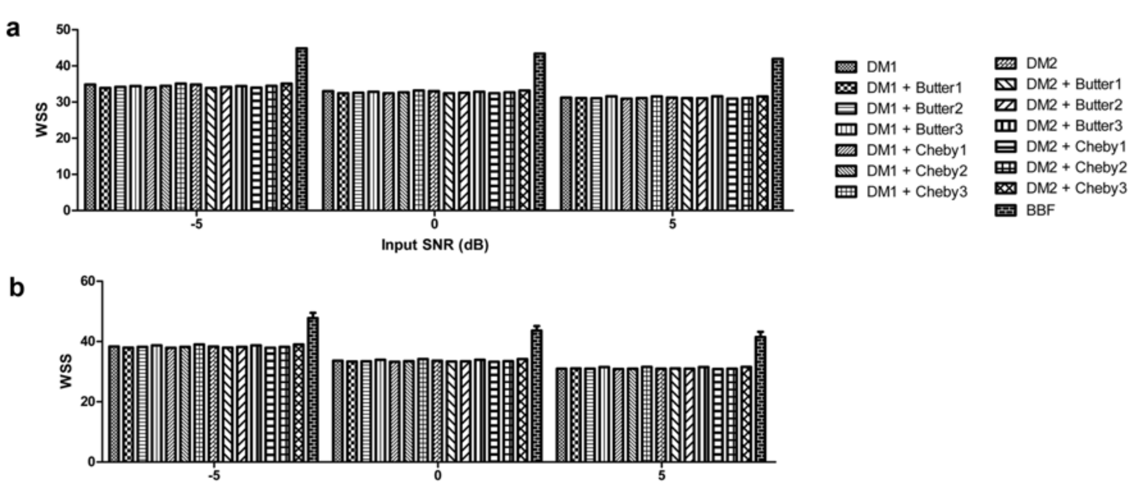

Fig. 10. The WSS index measurements for (a) speech sample 1 and (b) speech sample 2 in speech-in-white noise situations.

different from those in the speech-in-babble noise situations, but the trends were similar to those seen in the speech-in-babble noise situations.

DISCUSSION

In this study, we compared the effects of DM1- and DM2-

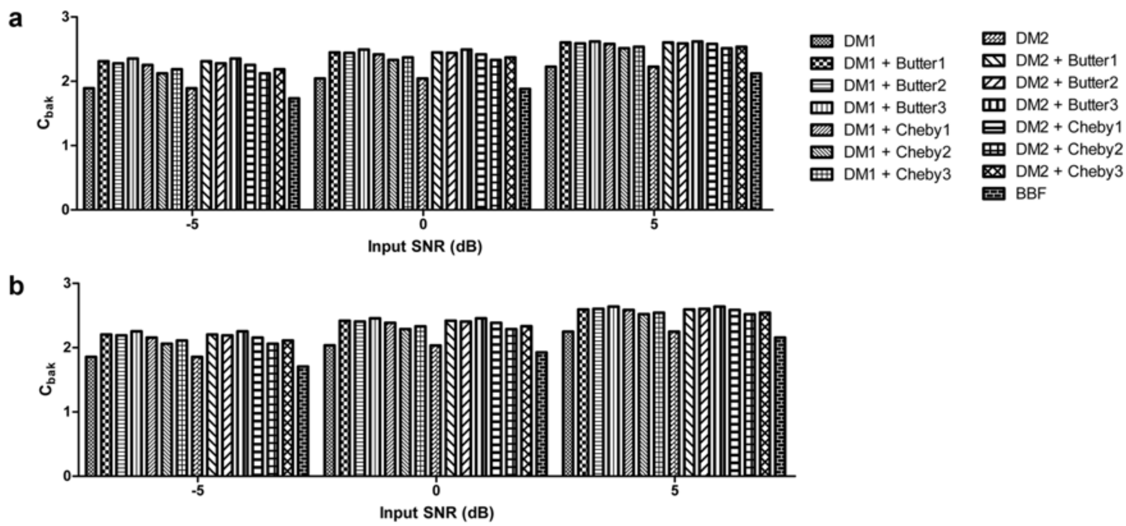

Fig. 11. The C_{bak} index measurements for (a) speech sample 1 and (b) speech sample 2 in speech-in-white noise situations.

based frequency-invariant beamforming algorithms and the BBF algorithm in a hearing aid situation by calculating four representative objective indices. The output SNRs of the DM-based algorithms were markedly improved by adding a compensating low-pass filter, but the degree of SNR improvement differed with the variations in filter type, filter order, kind of testing sentence, and input SNR conditions. In contrast, the other indices were not seriously affected by the use of the low-pass filter. The PESQ values were slightly higher for the BBF algorithm than the DM-based frequency-invariant algorithms, and the WSS values were slightly higher with the DM-based frequency-invariant algorithms than with the BBF algorithm. Adding low-pass filters to the DM-based algorithms did not result in significant enhancement of the PESQ and WSS values. We postulate that this is because these indices assume that the time delay is uniform in all frequency ranges during calculation, while the IIR filters used in this study have nonlinear phase characteristics. To acquire a more significant enhancement of the PESQ and WSS indices when adding IIR filters to the DM-based algorithms, it may be helpful to utilize IIR filters that have linear phase characteristics or to add an all-pass filter that can compensate for the phase distortion. In the case of C_{bak}, the attachment of a low-pass filter slightly improved the C_{bak} values; however, the degree of enhancement was not as great as the output SNR improvement because the value of C_{bak} was calculated by combining the PESQ, WSS, and segmental SNR values [15]. Generally, the values of PESQ, WSS, and C_{bak} were different in accordance with the types of environmental noises, and all of them could be improved by increasing input SNR conditions, but the improvements were not significant. Considering these experimental results, the performance of the DM-based frequency-invariant beamforming algorithms was superior to the BBF algorithm in a hearing aid situation. In addition, the implemented BBF algorithm had relatively long length of taps (512) which resulted in relatively long time delay (about 32 msec). Due to this time delay, in actual cases, the device user may experience deteriorated perception and to prevent this side-effect, the proper trade-off between the frequency-invariance and the time delay need to be investigated. Considering the computational burden of the algorithm, we recommended the application of first-order IIR filters to the DM-based beamforming algorithm to realize frequency-invariant characteristics of the beamformer for a real hearing aid system as there was no significant gain in performance enhancement by increasing the order of the IIR filters and the computational capacity and memory of hearing aid processors are very limited.

In this study, the performance of the trial algorithms was evaluated based on the assumption that there were no environmental reverberations, so that the effects of the beamforming algorithm alone could be observed. Therefore, sound data that were recorded in a non-reverberant environment were utilized in the experiments. However, the interference of speech and noise reverberation can affect the speech recognition ability of hearing-impaired patients in real indoor environments and, therefore, tests in a non-reverberant environment alone are not sufficient to evaluate the real effects of the trial algorithms in real hearing aid situations. Consequently, additional comparative evaluations of the performance of the trial algorithms in reverberant environments and further study to verify the real effects of the trial algorithms on hearing-impaired patients are required. In addition, in this study, only two kinds of environmental noises – a babble noise, which has similar characteristics to a speech signal, and a white noise, which has characteristics different from a speech signal – were utilized to observe the effects of environmental noises on the performance of the frequency-invariant beamforming algorithms. To more accurately evaluate the performance variations of the trial algorithms with different types of environmental noises, the same evaluation procedures will need to be repeated with more varied types of environmental noises, such as car, traffic, and wind noises. In addition, the performances of the trial algorithms were only evaluated *in vitro* in this study; however, the calculated values of each of the objective indices are not equivalent to the comfort and articulate feeling of hearing aid users, because the human auditory sense is affected not only by noise reduction but also by various additional psychoacoustic and cognitive factors. Therefore, in order to more realistically compare the clinical feasibility of the trial algorithms in hearing aid situations, additional clinical tests, such as the mean opinion score test and speech recognition threshold test, should be administered to a large number of hearing-impaired patients in hearing aid environments in order to evaluate the degree of improvement in speech intelligibility associated with these algorithms.

In this study, we used only the C_{bak} among the three commonly used composite indices (C_{sig}, C_{bak} and C_{ovl}). This is because the log likelihood ratio (LLR) index that is used to calculate C_{sig} and C_{ovl} is very sensitive to the phase distortion of the signal, as it models both the clean and testing signals as all-pole models. Therefore, when an IIR filter that has a nonlinear phase characteristic is attached to the DM-based algorithms, the values of LLR are worsened by the phase distortion. As a result, the values of C_{sig} and C_{ovl} are calculated abnormally and normal performance comparisons are not possible. Therefore, we excluded C_{sig} and C_{ovl} in this comparative study.

Recently, several beamforming algorithms for binaural hearing aids have been developed [16, 17] which utilize up to four embedded microphones (two in a right-side and two in a left-side hearing aid) to focus on four azimuthal directions of front, rear, left, and right (e.g., the auto ZoomControl of Phonak). In the binaural case, the polar patterns of the beamformer become distorted by the head-shadowing effect, and a spatial aliasing problem can occur because the distance between the two hearing aids is far longer than the wavelength of the speech signals, which can deteriorate the performance of conventional beamforming algorithms at high frequency ranges [18]. To evaluate the performance of conventional frequency-invariant beamforming algorithms in binaural situations, the trial algorithms need to be modified to fit to the requirements of binaural hearing aids by considering the head-shadowing and the spatial aliasing problems and by verifying the performance of the revised algorithms in binaural hearing aid situations.

CONCLUSION

In this study, the performance of various frequency-invariant beamforming algorithms in noisy environments was evaluated *in vitro* in a hearing aid situation using four objective indices. Our experimental results demonstrated that the DM-based frequency-invariant algorithm was superior in terms of SNR, WSS, and C_{bak}, and the BBF algorithm was superior in terms of PESQ. Considering these results and the computational burdens of the overall algorithms, we recommend using the DM-based frequency-invariant algorithm with a first-order compensation filter for real hearing aids. However, more *in vitro* evaluations using more input conditions and additional clinical evaluations, such as mean opinion score tests and speech recognition threshold tests, are required to more accurately verify the clinical feasibility of these algorithms.

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CONFLICT OF INTEREST STATEMENTS

Kyeongwon Cho declares that he has no conflict of interest in relation to the work in this article. Kyoung Won Nam declares that he has no conflict of interest in relation to the work in this article. Jun Chang Lee declares that he has no conflict of interest in relation to the work in this article. Sung Hwa Hong declares that he has no conflict of interest in relation to the work in this article. See Youn Kwon declares that he has no conflict of interest in relation to the work in this article. Jonghee Han declares that he has no conflict of interest in relation to the work in this article. Dongwook Kim declares that he has no conflict of interest in relation to the work in this article. Sangmin Lee declares that he has no conflict of interest in relation to the work in this article. In Young Kim declares that he has no conflict of interest in relation to the work in this article.

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