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Full Length Research Paper

Improving speech intelligibility of severely hearing-impaired people by frequency-lowering technique

Xianbo Xiao[1*], Guangshu Hu[1], Chunhong Liu[2] and Jia Liu[3]

[1]Biomedical Engineering, Tsinghua University, Beijing, 100084, China.
[2]Department of Otolaryngology, XuanWu Hospital, Beijing, 100084, China.
[3]Beijing Bo Na Feng Medical Instrument Co., Ltd, Beijing, 100028, China.

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Frequency lowering is an important audio signal processing method used in digital hearing aids. It helps patients with severe hearing loss in high frequencies by compressing speech into the patients' residual hearing regions. In this study, a general frequency lowering algorithm was proposed, which could separately process the contents in the disabled high frequencies and those in the low frequencies as well; and two different lowering strategies (the overlapped and segmented compression strategy) based on it were implemented, compared with traditional proportional frequency compression algorithm. The respective performances of both strategies were then evaluated using a specially designed software system, which simulated the signal processing procedure of digital hearing aids. After fitting this system to the patients, the results of a detailed speech intelligibility test showed that both of the frequency-lowering strategies provided more benefits, especially in consonant recognition, as compared with the hearing aids used by the patients. Furthermore, the segmented compression strategy showed better 'anti-noise' quality and was thus preferred by the patients. Finally, the impact of noise and training to the benefits of frequency lowering algorithms was also discussed in this paper.

Key words: Hearing aid, frequency lowering algorithm, speech intelligibility, compression strategy, software system development, noise.

INTRODUCTION

A hearing aid is an electro-acoustic body-worn apparatus, which transforms passing sounds to help the wearer achieve better hearing. The main purpose of hearing aids is to provide higher speech intelligibility among hearing-impaired individuals. Toward this end, the most popular algorithm used in hearing aids is the amplitude amplify-cation method. However, this has no remarkable benefit on the frequencies, because the sound power levels are elevated over the user's own hearing thresholds in every frequency. This is due to the fact that the HTLs (Hearing Threshold Levels) are near or over 120 dB. Furthermore, some studies have discovered that such method causes ill effects (Ching et al., 1998; Skinner, 1980; Turner,1999; Hogan and Turner, 1998; Goldbaum and Halpin, 1999). To solve this problem, the frequency-lowering technique was proposed, which transferred contents in the disabled frequency region into more sensible areas (Davis, 2001; Miller-Hansen et al., 2003; Parent et al., 1997; McDermont and Dean, 2000; Rees and Velmans, 1993). One simple example of this technique is playing back sound in a lower rate than its sampling rate. Here, although the band width is reduced to half, the contents are preserved over the whole band. Therefore, a woman's recorded voice sounding like that of a man's entails greater effort exerted in preserving speech features toward frequency lowering, instead of simply changing the output rate.

Two methods have already been employed toward lowering the speech spectrum. First is channel vocoding

*Corresponding author. E-mail: xianbo.xiao@gmail.com. Tel: (+86)10-84598233.

Table 1. Relevant information about the participants.

Listener	Sex	Age	Hearing impaired	Duration of hearing loss (yrs)	Own hearing aid type
L1	Male	27	Acquired	23	GnResound CANTA780D (2 yrs)
L2	Male	62	Acquired	26	Bernafon Smile+120 (3 yrs)
L3	Male	52	Acquired	16	Widex SD-9 BTE (2 yrs)
L4	Female	50	Acquired	10	Starkey ARIES (CIC) (5 yrs)
L5	Female	63	Acquired	28	Siemens MUSIC (4 yrs)
L6	Female	28	Acquired	26	Siemens MUSIC-P (3 yrs)

(Ling, 1969, 1967; Pinmonow, 1968; Takefuta, 1968), in which the high frequency speech contents are analyzed by a bank of filters whose output envelopes are used to control the amplitude of signals from low frequency synthesis filters. The second method is frequency transposition (Johansson, 1966), in which information in a specified high frequency band is shifted downward by amplitude modulation or nonlinear distortion. In this study, we propose a general frequency-lowering algorithm with an extremely flexible structure to implement both frequency compressing and shifting. This is accomplished by directly modifying the spectral coefficients in the frequency domain through Fourier Transformation (Patrick et al., 1983). This algorithm enables switching between the different compression strategies, and helps accelerate the fitting procedure through the considerate parameter settings. Using this algorithm, we have implemented two new frequency compression strategies, both of which processed the low frequency content by either overlaping or compressing. We have also established a software system to simulate the signal processing procedure of a digital hearing aid used to fit and evaluate the effect of the proposed algorithm during training and testing. By the end of the eight-week experiment, improvements over traditional hearing aids were found in both strategies.

MATERIALS AND METHODS

Frequency lowering algorithm and compression strategies

Frequency lowering algorithms: A frame of sound signal $\{x_n\}$ containing N data is discrete-Fourier-transformed (DFT) to a sequence of N complex spectral coefficients $\{X_k\}$ evenly distributed in the frequency band $[0, f_s]$, where f_s is the sample frequency. Every X_k can be expressed as follows:

$$\mathrm{Re}(X_k) = \frac{1}{N}\sum_{n=0}^{N-1} x_n \cos(\frac{2\pi nk}{N}); \quad k = 0,1,\cdots,N-1 \tag{1}$$

$$\mathrm{Im}(X_k) = \frac{1}{N}\sum_{n=0}^{N-1} x_n \sin(\frac{2\pi nk}{N}); \quad k = 0,1,\cdots,N-1 \tag{2}$$

Except those in $k = 0$ and $k = N/2$, all coefficients are conjugate symmetric with $k = N/2$. Therefore, there are only $(N/2)+1$ valuable coefficients, whose scales and phases can be presented as follows:

$$|X_k| = \left\{\mathrm{Re}^2(X_k) + \mathrm{Im}^2(X_k)\right\}^{1/2}; \quad k = 0,1,\cdots,N/2 \tag{3}$$

$$\theta_k = \frac{180}{\pi}\tan^{-1}\left[\frac{\mathrm{Im}(X_k)}{\mathrm{Re}(X_k)}\right]; \quad k = 0,1,\cdots,N/2 \tag{4}$$

If these $(N/2)+1$ coefficients are determined, we can infer all N coefficients according to the conjugate symmetry and obtain a real sequence after an inverse DFT.

Let us suppose that the frequency band $[0, f_s/2]$ is compressed into the region $[0, f_s/2c]$ at a compression ratio of c ($c \geq 1$), and $\{Y_k\}$ is used to denote the compressed spectral coefficients sequence. Then the compressed $N/2c+1$ spectral coefficients can be calculated by decimating from $\{|X_k|\}, k = 0,1,\cdots N/2$, and be relocated in the beginning of $\{|Y_k|\}$. Their phases are kept unchanged. In practice, the compressing ratio c could not always be an integer; thus the sequence $\{Y_k\}$ is represented as:

$$\begin{cases} |Y_k| = |X_{ck}|; \quad k = 0,1,\cdots,INT\left[\dfrac{N}{2c}\right] \\ |Y_k| = 0; \quad k = (INT\left[\dfrac{N}{2c}\right]+1),\cdots,\dfrac{N}{2} \\ \lambda_k = \theta_k; \quad k = 0,1,\cdots,\dfrac{N}{2} \end{cases} \tag{5}$$

In (5), $INT[(\cdot)]$ is the rounded-down integer of $(\cdot)$, and the values of $|X_{ck}|$ are determined by decimating the sequence $\{|X_k|\}$. Given that previous studies (Patent et al., 1997; McDermott and Dean, 2000) have indicated that intensive high frequency sounds will affect patients' hearing in low frequencies, coefficients in the individual patient's disable frequencies are set to be zeros. After manipulating the aforementioned frequency domain, we will deduce all the N coefficients according to the conjugate symmetry and obtain the time-domain sequence $\{y_n\}$ by inverse DFT. This process is the traditional proportional frequency compression algorithm characterized by a single important parameter, the compression ratio c.

The proportional method compresses the whole frequency band at a certain rate, disregarding the difference of the effects of the low and high frequency contents on speech perception, especially when the spectral contents are compressed in a narrower region. However, these are very important to patients with severely-impaired hearing. In this study, we propose a new algorithm that could separately process the contents in the disabled high frequencies and those in the low frequencies as well. At the same time, the compressed high frequency bands will be relocated somewhere in the original low frequency region.

If the frequency band $[f_L, f_H]$ is compressed by c and shifted to a region starting from f_R, we can obtain the transformed coefficients $\{S_k\}$ by:

$$\begin{cases} |S(f_R,f_L,f_H,c)_k| = |X_{c(k-N_R+N_L)}|; & k = N_R,\cdots,N_R+N_H-N_L \\ |S(f_R,f_L,f_H,c)_k| = 0; & k = 0,\cdots,N_R-1 \\ & \text{and } k = N_R+N_H-N_L+1,\cdots,\frac{N}{2} \\ \gamma_k = \theta_k; & k = 0,1,\cdots,\frac{N}{2} \end{cases} \tag{6}$$

where $N_R = INT\left[\frac{Nf_R}{f_s}\right]$, $N_L = INT\left[\frac{Nf_L}{c_H f_s}\right]$,

and $N_H = INT\left[\frac{Nf_H}{c_H f_s}\right]$.

Thus, the proposed algorithm can be represented by:

$$|Y(f_L,f_H,f_R,c_L,c_H)_k| = |S(0,0,f_L,c_L)_k| + |S(f_r,f_L,f_H,c_H)_k|; \quad \text{and } \lambda_k = \gamma_k = \theta_k; \quad k = 0,1,\cdots,\frac{N}{2} \tag{7}$$

where f_L, f_H, c_L and c_H represent the stop frequencies of the high frequency band and the compression rates of the low and the starting frequency of the relocated high frequency content, and λ_k stands for the phase value.

In this algorithm, the compression strategy can be customized by redefining the high frequency bands and retexturing the compressed contents, while preserving the significant spectral cues in the residual hearing region at the same time. Toward further understanding of the proposed algorithm, the two compression strategies are inferred by their distinct ways of processing the original low frequency content.

Two compression strategy

The overlapped compression strategy

Most of the energy and formant peaks in speech are located in the low frequency region. Thus, it is reasonable to believe that keeping the original low frequency information will provide a positive effect, leading to improvements in the patients' speech intelligibility. In this case, we changed the compression strategy by compressing the high frequency band and overlapping it on the higher end of the low frequency band. This was carried out while ensuring that the low frequency band will be left unaltered.

In (7), if $c_L = 0$, then the effective spectral coefficients processed by overlapped compression will be:

$$\begin{aligned} |Y(f_L,f_H,f_R,0,c_H)_k| &= |S(0,0,f_L,0)_k| + |S(f_R,f_L,f_H,c_H)_k| \\ &= \begin{cases} |X_k|; & k = 0,\cdots,N_R-1 \\ |X_k| + |X_{c_H(k-N_R+N_L)}|; & k = N_R,\cdots,N_R+N_H-N_L \\ 0; & k = N_R+N_H-N_L+1,\cdots,\frac{N}{2} \end{cases} \end{aligned} \tag{8}$$

This strategy is called the overlapped frequency compression, because of the overlapping that takes place in the region $k = [N_R, N_R+N_H-N_L]$.

The segmented compression strategy

In the overlapped compression strategy, some low frequency contents are mixed with the compressed high frequency contents. This could result in the loss of original low frequency information. We thus propose to separately compress the high and low frequency bands and place the compressed high frequency contents exactly behind the compressed low frequency contents. In this study, this will be called the segmented compression strategy.

In (7), if $f_R = f_L / c_L$, then the modified spectral coefficients in the segmented compression algorithm can be presented as:

$$\begin{aligned} |Y(f_L,f_H,f_L/c_L,c_L,c_H)_k| &= |S(0,0,f_L,c_L)_k| + |S(f_R,f_L,f_H,c_H)_k| \\ &= \begin{cases} |X_{c_L k}|; & k = 0,\cdots,N_R-1 \\ |X_{c_H(k-N_R+N_L)}|; & k = N_R,\cdots,N_R+N_H-N_L \\ 0; & k = N_R+N_H-N_L+1,\cdots,\frac{N}{2} \end{cases} \end{aligned} \tag{9}$$

and (9) will become

$$|Y(f_L,f_H,f_L/c,c,c)_k| = \begin{cases} |X_{c_L k}|; & k = 0,\cdots,N_H \\ 0; & k = N_H+1,\cdots,\frac{N}{2} \end{cases} \tag{10}$$

In specific conditions when $c_L = c_H = c$, we will have

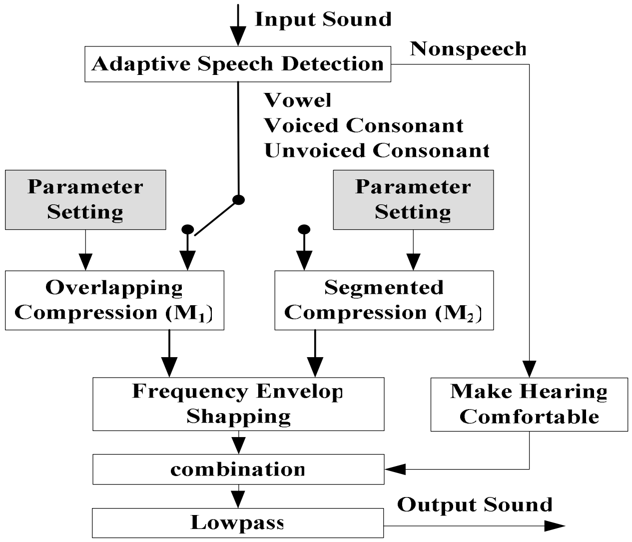

Figure 1. Block diagram of the software system used in this experiment, in which two frequency lowering strategies (the overlapped and the segmented compression) can be activated alternatively.

$N_R = N_L$, which actually performs proportional compression Compared with the traditionally used proportional compression strategy, the overlapped and the segmented compression strategies are more flexible for frequency component compressing and shifting. By adjusting the parameters in (7), the system employing this algorithm can transform the speech closer and produce a much more identical sound to what the patients expect.

For convenience and consistency, the variables M_1 and M_2 will be hereby used to represent the overlapped compression strategy and the segmented compression strategy, respectively. In general, M will stand for both of them.

Software system

To evaluate the frequency-lowering algorithms, a PC-based software system was first set up to simulate the signal processing procedure of a real digital hearing aid, as shown in Figure 1. The proposed frequency-lowering algorithm was incorporated in the system, and both strategies could be activated alternatively. The input sounds were sampled at 16k Hz and were processed in a 16 ms frame. An adaptive speech detection algorithm based on the spectrum envelope tracking technique (Marzinzik and Kollmeier, 2002) sent the frames to a 'making comfortable' branch or a more complex branch consisting of frequency lowering and envelope shaping (by NAL_RP formula). Afterwards, two branches were combined, and all frames were low-passed to produce an output. Except the frequency-lowering algorithm, all the programs in the system have passed real-ear wearing tests in a portable and digital hearing aid algorithms evaluation platform (Xiao et al., 2006) proposed in our former work.

Testing and training subjects

The six patients who participated in this study had hearing impairments that ranged from moderate to severe. As shown in Table 1, although they have different genders and age, all of them have had long histories of hearing impairment, and have had several years' experience of wearing traditional hearing aids. Moreover, all of the participants are Chinese speakers; hence all speech materials used during testing and training are recorded in Chinese Mandarin. Further details about the participants are presented below.

Testing and training procedure

In the experiment, the participants were provided with the proposed algorithm running two different strategies in the first and the second week. They were then asked to choose just one between the two algorithms and were instructed to use the one they preferred until the end of the third week. After five weeks of using their preferred algorithm, the participants were requested to undergo trainings with their chosen strategies. The trainings were conducted at least four hours a day, six days a week, during which the participants listened to and repeated the tape-recorded processed speech at home. The listening materials included clear speech, speech in various noisy conditions (e.g., in an office, in traffic, in a bus, in a restaurant), and other sound recordings (e.g., music, TV, and radio sounds). Once a week, the listeners took the hearing tests and the speech recognition tests. In addition, a survey was conducted among the participants in order to obtain an objective evaluation of the proposed algorithm and the two strategies employed.

All the fitting and testing procedures were performed in a quiet room in a hospital, and the speakers were positioned one meter in front of the participants. The testing sounds were broadcasted in 60dB, regardless of whether or not external noise was present. The clear speech recordings used in this study were recorded following the previous works presented (Cai, 1963; Chen, 1966). In other studies (Turner and Hurtig, 1999; Sakamoto et al., 2000), it was noted that the male and female speakers influenced the frequency lowering of speech for the listeners with sensor neural hearing loss. Therefore, for this study, we used a female voice to avoid confusion.

RESULTS

Hearing threshold levels

The threshold levels of the participants, aided by the M algorithms and by their own hearing aids, are shown in Figure 2. As can be seen, the HTL curves of the M and the OA (Own hearing Aid) below 2k Hz are quite similar. This could be attributed to the fact that the frequency lowering program was not yet activated at that time, and the envelope shaping (if it was chosen to work) was in charge. When frequency lowering was activated, the $M_1$and the M_2 curves increased much more than the OA curves. This indicated that the listeners obtained more improvements in the M algorithms by regaining information from the disabled high frequencies.

Speech intelligibilities

Given that speech intelligibility improvement is the main purpose of hearing aids, this factor has always been used to evaluate the performance of relevant algorithms. Fi-

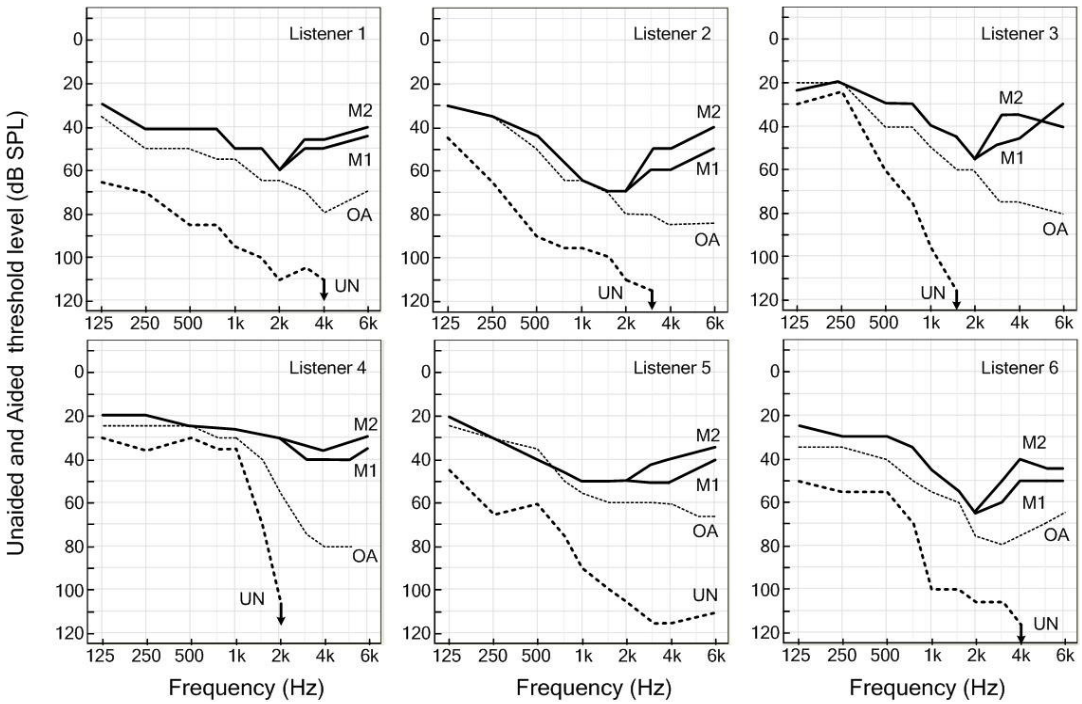

Figure 2. Hearing threshold levels testing results of the six participants in an unaided and aided situation, as measured with narrow-band noise. Here, UN refers to "unaided", OA refers to "aided with listeners' own conventional hearing aids", M_1 refers to "aided with M_1 algorithm", and M_2 refers to "aided with M_2 algorithm". The arrows indicate that hearing thresholds beyond these frequencies are immeasurable.

gure 3 illustrates the result of a detailed speech intelligibility test taken by the participants at the end of the third and eighth weeks (in red, only with the chosen algorithm). Likewise tested was the 5 dB Gaussian white-noise added speech (right 3 panel of Figure 3). This was considered because many complaints about noises have been mentioned in previous studies.

As illustrated in Figure 3, the M algorithms obtained better scores than the users' own hearing aids in most tests, particularly in the consonants recognition test. This advantage further increased in the next five weeks, regardless of which algorithm was chosen by the participants. The results in the test conducted after the third week is consistent with the participants' selection at the end of the third week; hence, all users, except L6, chose M_2 algorithm in the continuation of their experiments. In addition, M_2 showed better performance than M_1 in most tests.

Upon adding noise, the accuracy rates of the M algorithms decreased greatly, even to a lower value than that for OA's recorded period. However, the deviations of the accurate recognition rates increased. These indicated that the initial benefit received from the M algorithms became increasingly unstable under noise-filled conditions. After the five-week training, a decrease was also observed; however, such decrease did not affect the M algorithm's superiority over traditional hearing aids. It was also found that the decrease of M_2 scores was a little less than that for the scores of M_1 in most cases. This could indicate that M_2 tended to be steadier than M_1 in noisy conditions.

Performance in various environments

In the final survey, the participants rated and compared the respective performances of both M algorithms with those of the traditional hearing aids with regard to processing speech in various environments. The average scores presented in Figure 4 illustrates the fact that the M algorithms performed much better in quiet environments (e.g., in an office), while no remarkable advantages could be found in speech heard in a noisy environment (e.g., in a restaurant). The large deviations in the restaurant and TV situations indicated the benefit of the M algorithms, such that it tended to be unstable and sensitive to the environment and input speech quality, and would even lead to inverse effect at some point. Moreover, in the survey, almost all participants agreed on the benefits obtained within the traffic environment, which might be caused by the relative depression of the original low frequency content where automobiles' hoot sounds inhabited.

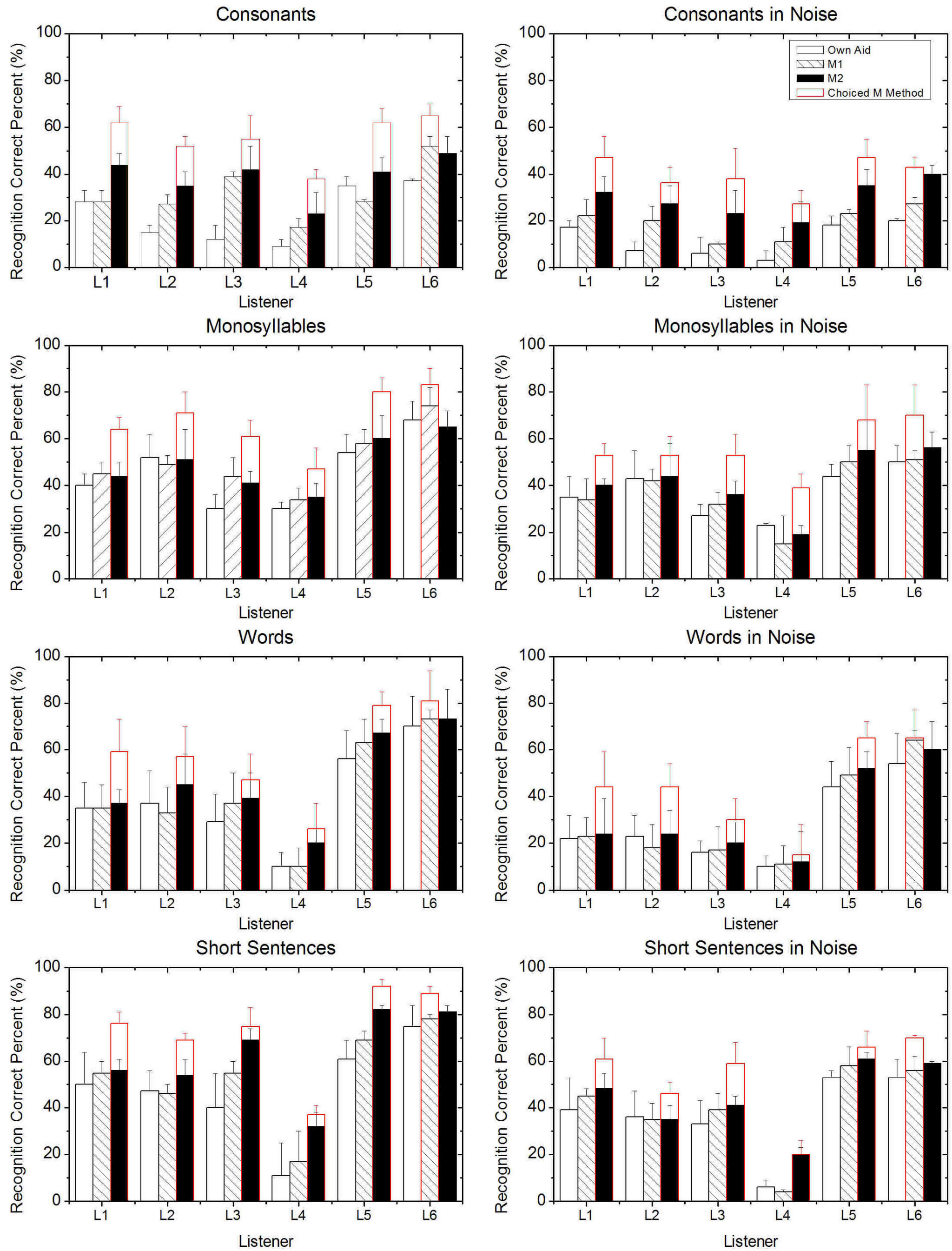

Figure 3. Results of the speech intelligibility tests taken by the six participants (L1~L6) at the end of the third and eighth weeks (red). The variables included in the test include the Chinese consonant, monosyllables, words, and short sentences.

DISCUSSION

Effect of training

Most listeners were unfamiliar or even felt some degree of discomfort during their first encounter with the frequency lowering technique. They perceived it to be 'unnatural' and 'noisy'. Therefore, as supported by previous studies, a well-spent period of training would be helpful for the participants to be more accustomed to the new method. In this study, they were requested to undergo a five-week training, during which results of the short

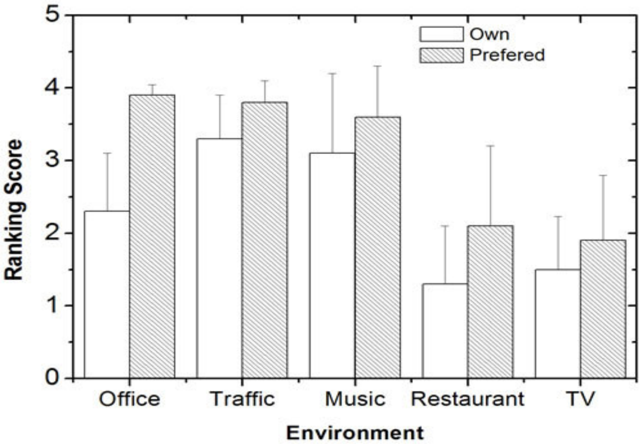

Figure 4. Average performance ranking scores of the participants' own and preferred frequency lowering algorithms in various environments. The participants were requested to rank them from 0 to 5, with regard to ease of communication, background noise factors, and the overall pleasant quality of hearing they experienced.

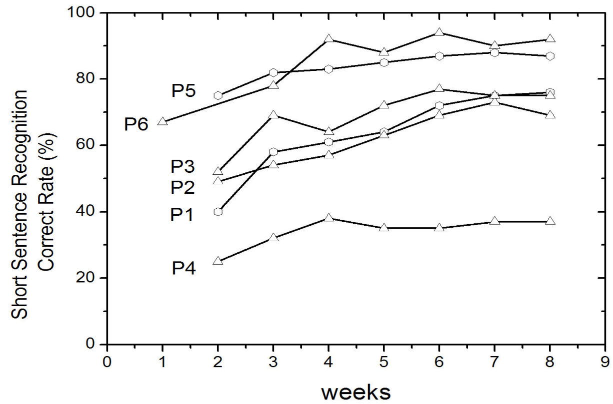

Figure 5. Short sentence recognition accuracy rates of the six participants with their preferred frequency lowering algorithms, as tested at the end of each week of training.

short sentence recognition tests taken every week were recorded to observe the effects of such training (Figure 5). Since L5 chose M_1, her first score was recorded after week 1. It was found that her speech intelligibility level kept increasing during the first two to three weeks and then settled to a stable level in the next three to four weeks. This indicated that with properly arranged training, it is possible for the users to be accustomed to M algorithms within six weeks.

Comparison of the two strategies

As presumed, by employing different strategies of processing original low frequency content and relocating high frequency content, the proposed algorithm showed remarkable differences in the hearing and speech intelligibility tests taken by the participants, thereby justifying their preference for the proposed algorithm.

The overlapped compression strategy (M_1) accomplisshed a significant performance in preserving the nature of the original low frequencies by allowing mixing in a restricted region. If a patient considers hearing comfort and sound quality more than the other factors, M_1 might be an appropriate choice. On the other hand, the segmented compression strategy (M_2) compressed both the low and high frequency regions to keep them from overlapping. This would possibly cause a decrease in sound quality, but could still provide higher speech intelligibility. Thus, more participants preferred this algorithm over the other one used in this study. Performance in a noisy condition is another important feature that should be considered in designing algorithms applied in hearing aids.

If the input sound contains noise, the noise will keep its level when compressed, but will be piled up when overlapped. The increase of relative noise power may cause damage to the users' speech recognition because of their poor resistance to noise. In this aspect, M_2 was considered to have better 'anti-noise' quality than M_1, as shown in the decrease of scores noted during the noisy condition tests.

Conclusions

In this study, we proposed a general frequency-lowering algorithm and evaluated its performance through real-ear fitting and testing. The proposed algorithm was explicit in theory and could be switched between the different lowering strategies. In the speech intelligibility tests, the system employing the proposed lowering algorithm showed superior performance over the traditional hearing aids.

The results also indicated that different lowering strategies were required when fitted to the individual patient. It should also be noted that different strategies have their respective features. In view of this, future research may endeavour to focus on realizing the optimal frequency-lowering strategy in an online system, as well as a further benefit investigation in a bigger patient group.

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