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Evaluation of frequency-lowering algorithms for intelligibility of Chinese speech in hearing-aid users

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Abstract

Despite the attention being given to and the knowledge of the benefits of evaluating frequency-lowering algorithms for hearing-impaired people, the causality between these algorithms and their benefits is still not clear. This is aggravated by the fact that comparative research on the methodologies and skills required for fitting an appropriate algorithm to individual patients is lacking. Against this backdrop, the current study has attempted to make progress in this area. In this experiment, six experienced traditional hearing-aid users with severe impairments in the high frequencies were fitted with two different frequency-lowering methods, and weekly hearing tests were conducted to track the benefits of such methods. After the experiment, five of the listeners accepted the frequency-lowering algorithms. Both methods showed superior results when compared with the listeners' own hearing aids in most of the tests, and the segmented compression algorithm was indicated to have better "anti-noise" quality and speech intelligibility improvement capabilities. A preference for the application of proportional compression to unvoiced speech was also found in this algorithm. Unlike in previous studies, all speech materials here are recorded in Chinese. Therefore, the results could also be used to evaluate the benefits of frequency-lowering to the intelligibility of Chinese speech for patients. As a result, an improvement was found, especially in the recognition of consonants. Moreover, no adverse effect was found in intonation recognition.

Keywords: Hearing aid; Frequency-lowering; Speech intelligibility

1. Introduction

A hearing aid is an electroacoustic bodily worn apparatus which transforms passing sounds to help the wearer achieve better hearing. Higher speech intelligibility is the principal target, and amplitude amplification is the algorithm used in hearing aids to achieve such an end. However, this kind of method, which elevates the sound power levels over the users' own hearing thresholds in every frequency, has no remarkable benefit on the frequencies where the hearing threshold levels (HTLs) are near to or over 120 dB. In some studies, it has even been found to have detrimental effects [1–5]. To solve this problem, frequency-lowering has been proposed, whereby the frequency of the sounds from the disabled region is shifted into more sensitive frequency regions [6–10]. One simple example of frequency-lowering is the playing back of sounds at a lower rate than at the sampling rate. Although the bandwidth is halved, the contents over the whole band are preserved. In this case, a woman's recorded voice would sound like that of a man, requiring a significant amount of work to preserve the speech features in the process of frequency-lowering instead of simply changing the output rate. Algorithm design and parameter setting result

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in the corresponding reactions in sound quality and speech intelligibility; therefore, a better understanding of the causality among them would be helpful in algorithm optimization and fitting efficiency.

In this study, we investigated the effects of two frequency-lowering algorithms for high-frequency hearing-disabled patients. Much attention has been paid to associating the differences in the effects with the changes in lowering strategies and parameter settings by conducting a comparative study. Hearing tests were conducted weekly, with three weeks allotted to fitting and five weeks to training. According to the results, the variation in the application of overlapping and compression in the frequency domain was related to the algorithms' performances in terms of the hearing comfort, noise resistance, and speech intelligibility improvement. The comparison of the proposed two algorithms is discussed. Furthermore, because few previous studies have been based on Chinese speech, it was chosen as the language in all the training and testing materials. As we had anticipated, frequency-lowering showed benefits to most listeners in this study, and no adverse effect was found on intonation recognition. In addition, the results of further tests are discussed for evaluating in detail, and so as their benefit to Chinese speech recognition. The effect of training and the comparison between the two proposed algorithms are also discussed.

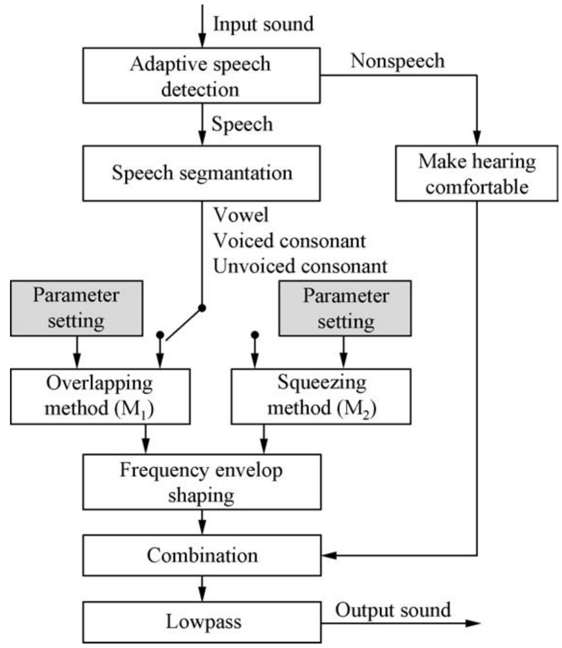

Fig. 1. A diagram of the software system used in this experiment, in which two frequency-lowering methods (overlapped and segmented compression) can be activated alternatively.

2. Materials and methods

2.1. System and algorithms

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 Fig. 1. Both frequency-lowering algorithms were incorporated into it and were activated alternately. The input sounds were sampled at 16 kHz and processed in a frame of 16 ms. An adaptive speech detection algorithm based on the spectrum envelope tracking technique [11] sent non-speech frames for "making the listener comfortable", or a more complex processing composed of frequency-lowering and envelope shaping [12] (using the NAL_RP formula [13]). All frames were then combined and were low-passed in terms of their output. All programs in the system, except for the frequency-lowering methods, have passed real-ear hearing tests in a portable digital hearing aid algorithm evaluation platform [14], based on our previous work.

There are two methods that have been employed for lowering the speech spectrum: one is channel vocoding [11,15–18] 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 other one is frequency transposition [19] in which information in a specified high-frequency band is shifted downward by amplitude modulation or nonlinear distortion. Both frequency-lowering algorithms investigated in this paper are based on the latter because of its comprehensibility and practicality. These two algorithms compress and shift the high-frequency contents in the same manner, but relocate them with original low-frequency contents in a different way.

An example of phoneme /s/ is given in Fig. 2 to show how the two methods shift the frequency-domain content using different lowering strategies. Apparently, in Fig. 2b and Fig. 2c, both algorithms accomplished the same mission: compressing the contents from 0–8 kHz to 0–4 kHz. However, in Fig. 2b, the compressed high-frequency contents are overlapped and mixed with the original contents at 3–4 kHz, keeping the lower frequencies undisturbed. This method is a typical downward shift scheme of frequency-lowering. One such scheme was developed in Ref. [20] and was followed by many studies carried out on similar and improved algorithms [21,22]. Meanwhile, in Fig. 2c, the low-frequency contents are compressed as well to make room for the high-frequency contents, which introduced distortion in low frequencies but avoided overlapping. This method is born from proportionate frequency compression, in which spectral information shifts all frequency components downwards by a constant factor. Some listeners have obtained speech perception benefits when listening to proportional frequency compression [23]. It is expected that the different compression and relocating methodologies would cause differences in the nature of the output sound and the performance in noisy environments, which are major concerns for hearing-aid users. Therefore, understanding the results caused by the differences between algorithms is expected to help increase an

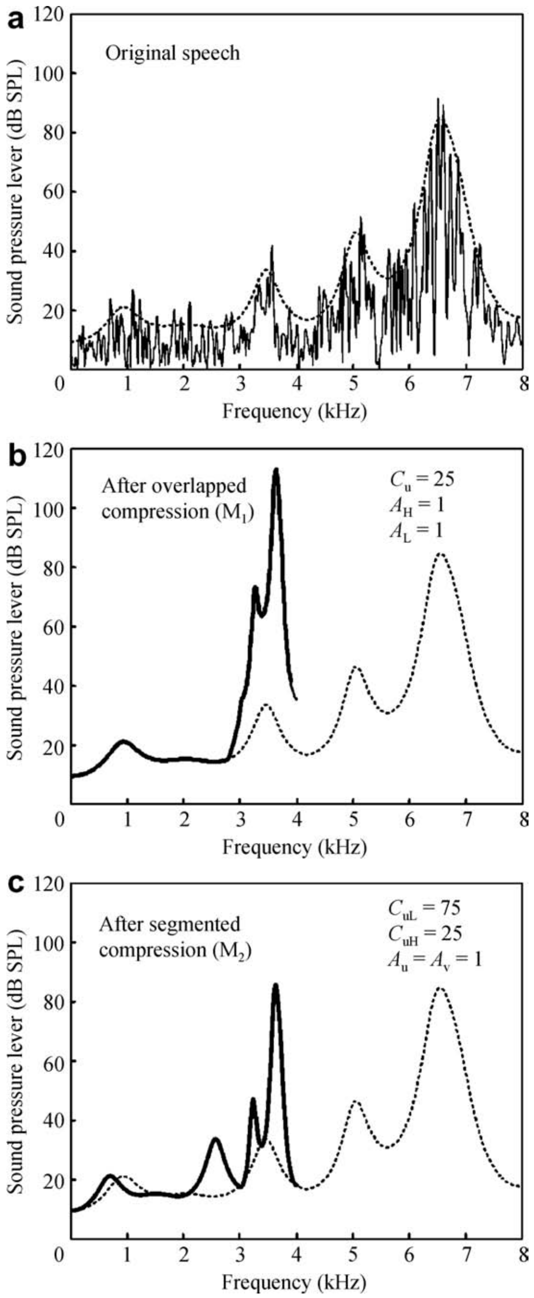

Fig. 2. Original spectral envelope of /s/ in the word "speaker" spoken by a man (a), and what it looks like when its high-frequency contents are lowered by the overlapped (b) and segmented compression methods (c). Corresponding parameters labeled in each panel are legible intuitively, but are not necessarily appropriate in practical fitting.

individual's acceptance by allowing the appropriate algorithm to be fitted.

For convenience, M_1 and M_2 were used to represent the overlapped compression algorithm in Fig. 2b and the segmented compression algorithm in Fig. 2c, respectively.

2.2. Parameter setting and fitting

Adjustable parameters were used in this study to regulate the system's running and algorithms' fitting. The parameter settings followed the design in Ref. [15]. The main adjustable parameters are listed below.

(1) Compression coefficients: C_{uH}, C_{vH} in M_1 and C_{uL}, C_{uH}, C_{vL}, C_{vH} in M_2 (the subscripts 'u', 'v', 'L', and 'H' stand for unvoiced speech, voiced speech, low-frequency region, and high-frequency region, respectively). The compression coefficient is defined as $C = BW_{\text{output}}/BW_{\text{input}} \cdot 100\%$, ranging from 10 to 90 in steps of 10.
(2) Gain coefficients: A_{uH} and A_{vH} in dB SPL, added to the compressed high-frequency contents after shifting. These parameters could regulate the intensity ratio between peaks in the high- and low-frequency regions. An inappropriate ratio might markedly deteriorate the algorithm's efficiency according to Ref. [17].
(3) Mode selector: this parameter controls the fundamental frequency recovery R ($R = P_{\text{out}}/P_{\text{int}} \times 100\%$, only in M_2 algorithm) and the switch of the frequency-lowering algorithms. In mode 1 $R = 100$, in mode 2 $R = C_{vL}$, in mode 3 $R = C_{vH}$, and in mode N, the frequency-lowering program is shut down.
(4) Envelope shaping switcher: this is used to turn the frequency shaping program on and off.

The above-mentioned parameters should be fitted in the order of 'voiced speech related', 'unvoiced speech related', 'mode selector', and 'envelope shaping switcher'. This is consistent with their importance in terms of the algorithms' effect. The whole fitting procedure was carried out under the direction of audiologists. They listened to a patient's description and helped to adjust the parameters according to the subject's reaction.

2.3. Subjects

Six moderate to severely hearing-impaired listeners participated in the experiment. They were of different gender and age, but all had had a long history of hearing impairment and several years of experience in wearing traditional hearing aids. All the subjects were Chinese speakers, and they gave their informed consent to participate in the experiment. All speech materials used in the testing and training were recorded in Mandarin. More details about the listeners are listed in Tables 1 and 2.

2.4. Procedure

The proposed system and algorithms were first evaluated in normal hearing people. No obvious rejection was reported and some listeners showed improvements in speech perception. But according to the previous studies, effective frequency-lowering algorithms could provide quite varied effects in normal hearing people. So the results of an experiment on qualified patients will be most reliable.

In the experiment, the listeners were fitted with hearing aids using two proposed algorithms in the first and second

Table 1
Relevant information about the listeners.

Listener	Gender	Age	Hearing impaired	Duration of hearing loss (yrs)	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)

Table 2
Listeners' unaided pure-tone hearing thresholds in dB HL.

Listener	Frequency (Hz)										
	125	250	500	750	1000	1500	2000	3000	4000	6000	8000
L1	65	70	85	85	95	100	110	105	110	–[a]	–
L2	45	65	90	95	95	100	110	115	–	–	–
L3	30	25	60	75	95	115	–	–	–	–	–
L4	30	35	30	35	35	70	105	–	–	–	–
L5	45	65	60	75	90	100	105	115	115	110	100
L6	50	55	70	100	100	105	105	105	115	–	–

[a] '–' means no response could be elicited at the frequency by the audiometer used.

weeks, and they were free to use them until the end of the third week when they were asked to choose only one. In the following five weeks, the listeners were requested to take training with their chosen algorithms for at least 4 h per day, 6 days per week, by listening to and repeating the processed speech recorded on tapes at home. The listening materials included clear speech, speech in various noisy conditions (such as in an office, during traffic jams, on a bus, in a restaurant), and other sound recordings (for example, music, TV and radio sounds). Once in a week, the listeners took hearing tests and speech recognition tests. A survey was conducted at the end of this period in order to allow an objective evaluation of both algorithms to be made. Our analysis, discussion, and conclusions are based on the data acquired from the above-mentioned tests.

All the fitting and testing processes were accomplished in a silent room in a hospital, with the speakers placed 1 m in front of the listeners. The testing sounds were broadcasted at 60 dB regardless of the level of background noise. The clear speech recordings used in this study were recorded following previous work [24–27]. Some studies [15,23] have already identified the influence of male and female speakers on the frequency compression of speech for listeners with sensorineural hearing loss, so we used only the female voice in the current work.

3. Results

3.1. Parameters

The fitting results are listed in Table 3. As shown, five of the six listeners accepted the frequency-lowering algorithms after the experiment. The results also indicate the following:

(1) The high-frequency compression coefficients between the two algorithms were close.
(2) Higher gains were applied to the compressed high-frequency contents in M_1 compared with M_2. They were probably used to meet the greater demand of reducing interference from the original information in the overlapping region.
(3) There was less compression in the low-frequency region, consistent with the conclusions of the previous studies [15,16].

Table 3
Listeners' final parameter settings after fitting.

Listener	Mode	Overlapped compression (M_1)						Segmented compression (M_2)							Choice after three weeks	Final acceptance
		C_{vH}^{-1} (%)	C_{uH}^{-1} (%)	A_{vH} (dB)	A_{uH} (dB)	Envelope shaping	Mode	C_{vL}^{-1}(%)	C_{vH}^{-1}(%)	C_{uL}^{-1} (%)	C_{uH}^{-1} (%)	A_{vH} (dB)	A_{uH} (dB)	Envelope shaping		
L1	1	30	40	4	5	ON	3	80	30	40	50	4	4	ON	M_2	M_2
L2	1	20	40	4	3	ON	3	60	40	30	30	3	3	ON	M_2	M_2
L3	1	30	50	4	4	ON	2	70	20	40	40	3	3	ON	M_2	M_2
L4	1	40	50	3	6	OFF	2	80	30	30	30	3	8	OFF	M_2	–
L5	1	40	40	3	6	ON	1	70	30	30	30	4	5	OFF	M_2	M_2
L6	1	20	40	3	9	ON	3	70	30	40	40	2	7	ON	M_1	M_1

(4) In M_2, when processing unvoiced speech, almost similar compression coefficients were set to low and high frequencies. This indicated the listeners' preference in applying proportional compression (compressing the whole frequency band at a certain rate [6]) to unvoiced speech in M_2.

All the above conclusions were again exhibited in the results of subsequent tests. They could be explained by the user's hearing demand; thus, a connection between the parameters and the algorithm's performance was established.

The results of the hearing test and speech intelligibility test are discussed in the following section.

3.2. Hearing threshold levels

The threshold levels of those unaided, those aided with M algorithms, and those aided with their own hearing aids are shown in Fig. 3. The HTL curves of M and OA (users' own hearing aids) below 2 kHz are quite similar because the frequency-lowering program was not activated, and the envelope shaping (if set to ON) was in charge. When the frequency-lowering was activated, the M_1 and M_2 curves rose more than the OA curves. This indicated that the listeners had more improvements in the M algorithms by regaining information from the disabled high frequencies.

3.3. Speech intelligibilities

Improving speech intelligibility is the main goal of hearing aids, so it is always used to evaluate the performance of relevant algorithms. Fig. 4 shows the results of a detailed speech intelligibility test taken at the end of the third and eighth weeks (bars in red indicate the results obtained only with the chosen algorithm). A 5 dB Gaussian white-noise added to speech was also tested (on the right panel of Fig. 4) because of the many complaints found in the previous studies in relation to noise.

As shown in Fig. 4, the M algorithms resulted in better scores than the users' own hearing aids in most tests, especially in the consonants recognition test. This advantage increased further in the following five weeks, regardless of the algorithm used. The testing results after the third week were consistent with the listeners' selection in Table 3, in which all users except L6 chose M_2 as they continued with the experiments. In comparison, M_2 showed better performance than M_1 in most tests.

After noise was added, the accuracy rates of the M algorithms decreased greatly, even to a value lower than that of the OA's. The deviations in accurate recognition rates increased too. These indicate that the M algorithms became more unstable in noisy conditions. After five weeks of training, a decrease also occurred, but this time, it affected the M algorithms' superiority over traditional hearing aids. It was also found that the decrease in M_2

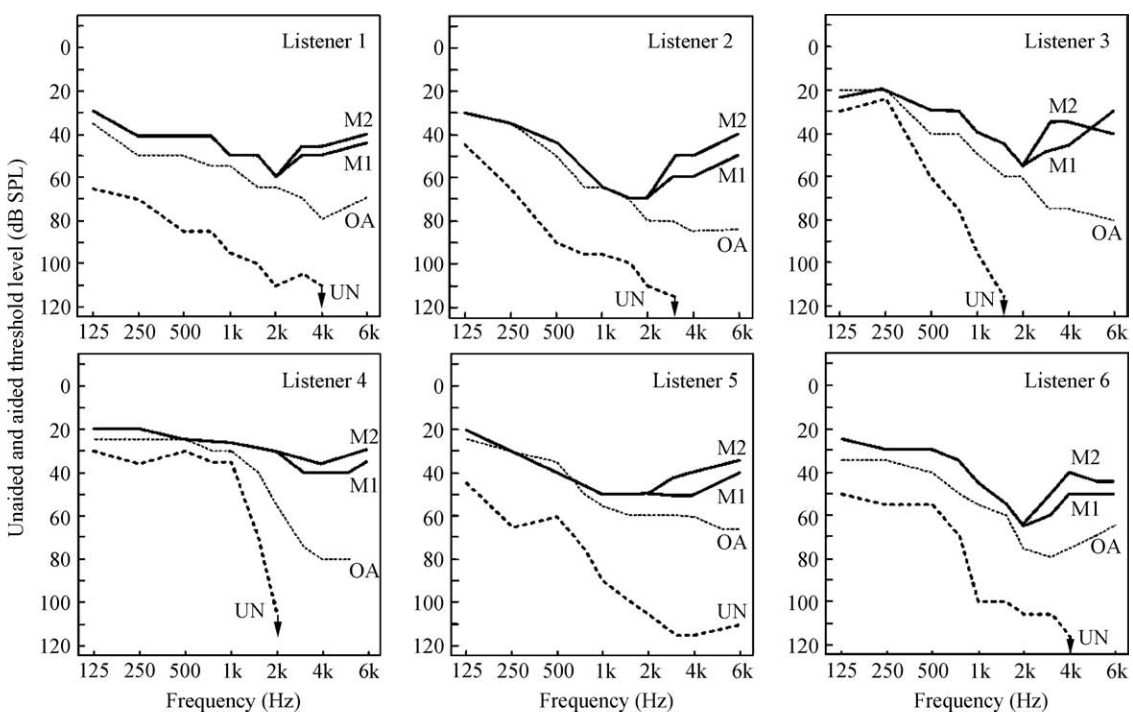

Fig. 3. Hearing threshold levels of the six listeners in unaided and aided situations, measured with narrow-band noise. UN, unaided; OA, aided with listeners' own conventional hearing aids; M_1, aided with M_1 algorithm; and M_2, aided with M_2 algorithm. The arrows indicate that the hearing thresholds beyond these frequencies are immeasurable.

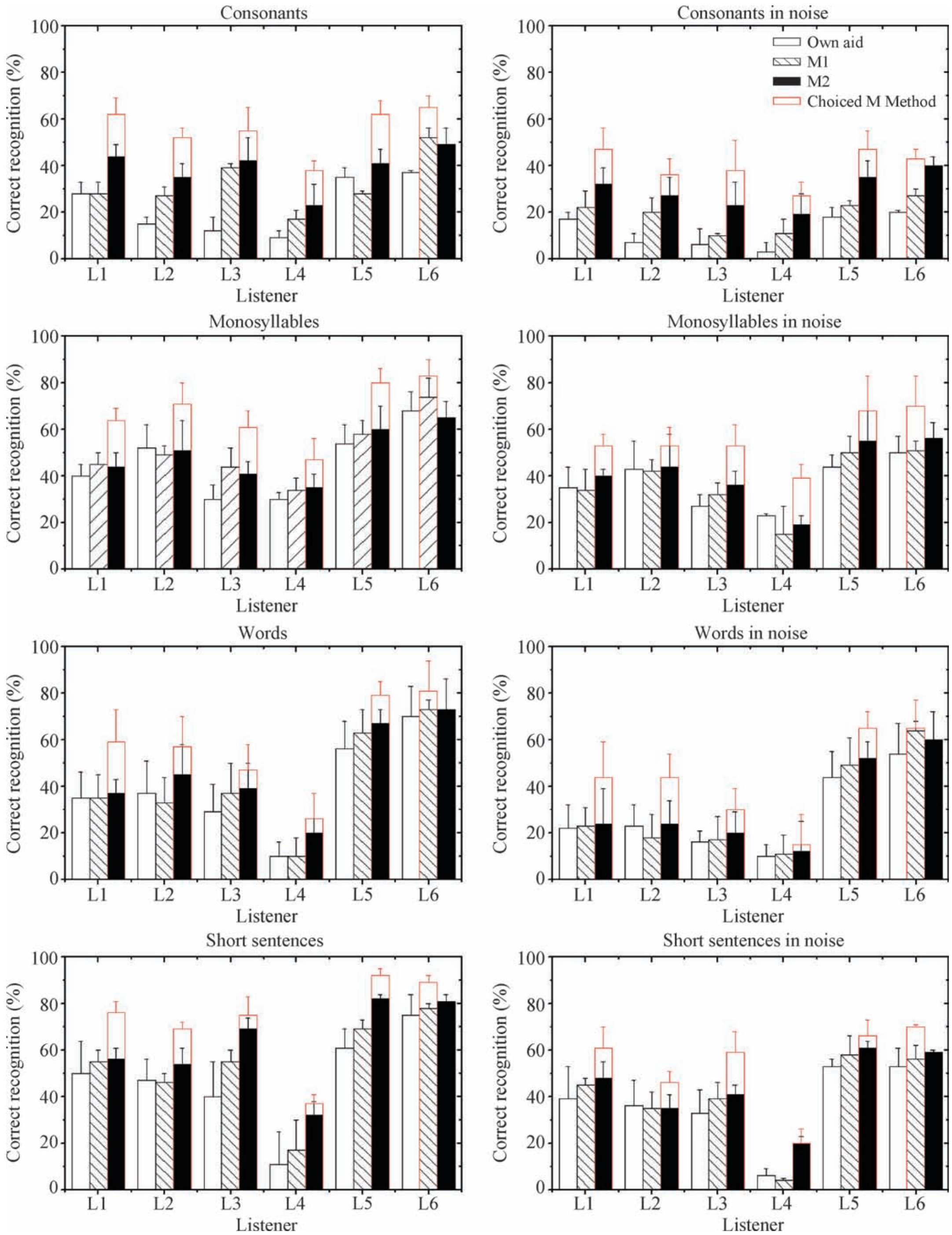

Fig. 4. Speech intelligibility test results of six listeners (L1–L6) for Chinese consonants, monosyllables, words, and short sentences taken at the end of the third week and at the end of the eighth week (red).

scores was a little less than that of M_1 scores most of the time, which might indicate that M_2 tended to be steadier than M_1 in noisy conditions.

Chinese speech is different from English speech in many aspects. Unlike English, Chinese has no allophone of consonants. Moreover, Chinese has its own distinct features such as intonation, syllable, tone-sandhi, light tone, and retroflex finals. Therefore, a more detailed recognition test in the phoneme level was taken in this study. The averaged comparative results of phoneme recognition rates between the chosen algorithms and the listeners' own hearing aids are shown in Fig. 5. Thirty-five finals and nineteen initials were utilized in this test. Most people from South China had difficulty in distinguishing blade-alveolars /z/, /c/, /s/ and corresponding blade-palatals /zh/, /ch/, /sh/, so we calculated their scores in pairs, as shown in Fig. 5c.

As a result, most scores in Fig. 5 were beyond zero, which indicated that frequency-lowering algorithms were

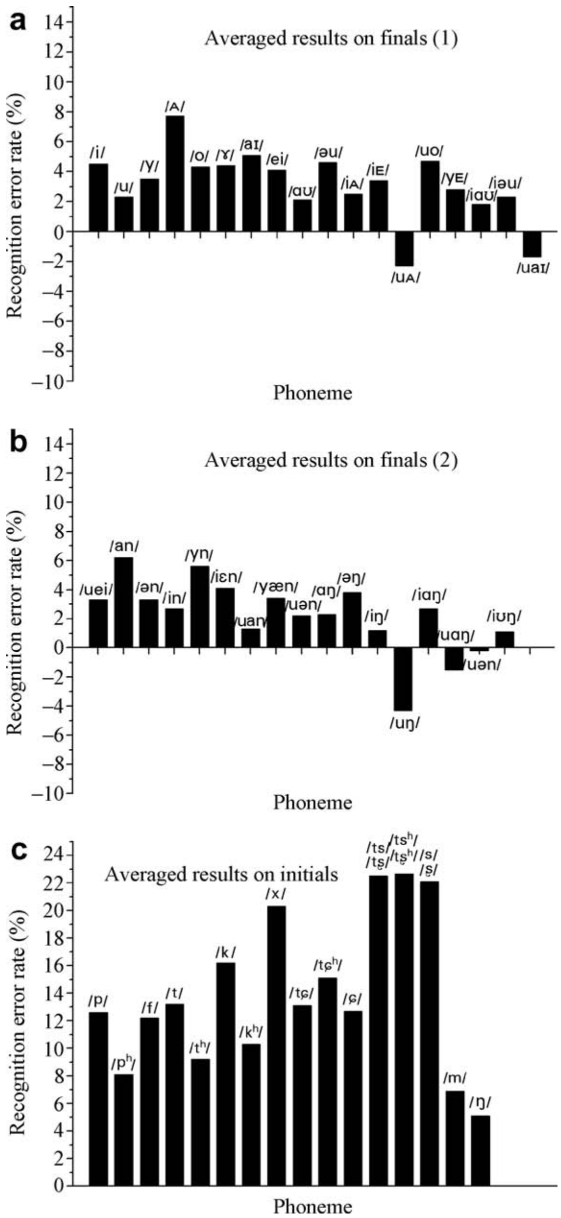

Fig. 5. Averaged comparative accuracy rates for the Chinese phoneme (labeled in IPA symbol [28]) recognition test between listeners' chosen frequency-lowering algorithms and traditional hearing aids. A positive value indicates that frequency-lowering is superior and vice versa.

superior to traditional hearing aids, especially in the recognition of initials, that is, with 7–18% advantage. However, the benefits obtained from the finals were subtle. An inverse effect was even found for some diphthongs and triphthongs.

Intonation is most important in Chinese speech perception, so a recognition test of intonation was taken at the end of the experiment. Monosyllables and words were played by speakers, and listeners were requested to identify the intonation in 3-s intervals. Correct recognition rates, as shown in Fig. 6, indicated that frequency-lowering had no negative effect on intonation recognition.

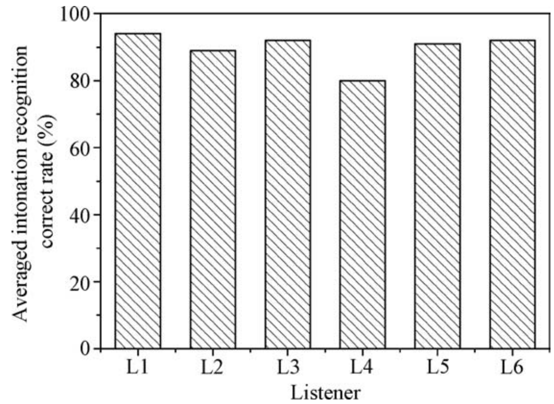

Fig. 6. Averaged intonation recognition correct rates for six listeners.

3.4. Performance in various environments

In the final survey, the listeners were scored to compare the performances of the M algorithms with traditional hearing aids in terms of processing speech in various environments. The averaged scores are shown in Fig. 7. M algorithms performed much better in a quiet environment, for example, in the office, while no remarkable advantage was found in speech with a noisy background, for example, in a restaurant. The large deviations in restaurant and TV situations indicate that the benefit of M algorithms was unstable and sensitive to the environment and input speech quality. Sometimes, it would even lead to an inverse effect. Moreover, in the survey, almost all listeners agreed on its benefit when used during a traffic jam. This might be caused by the relative depression of the original low-frequency content where automobile hooting sounds were included.

3.5. Effect of training

At first, most listeners were unfamiliar with and even felt uncomfortable when experiencing frequency-lowering.

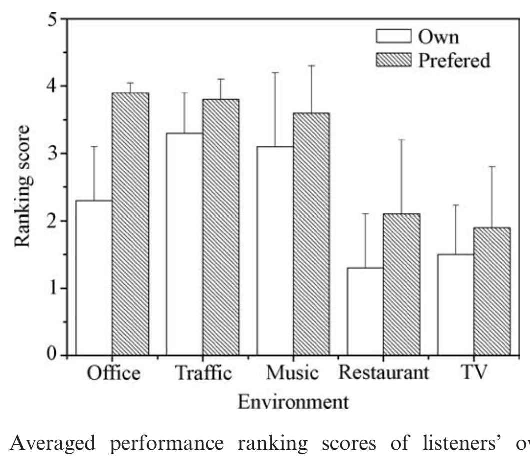

Fig. 7. Averaged performance ranking scores of listeners' own and preferred frequency-lowering algorithms in various environments. Listeners are requested to rank them from 0 to 5, considering ease of communication, background noise interference and comfort of hearing.

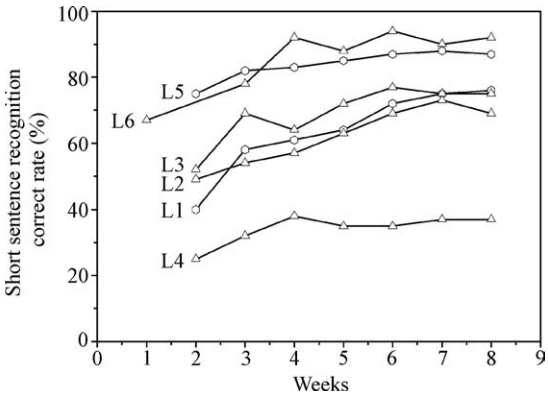

Fig. 8. Short sentence recognition accuracy rates of six listeners with their preferred frequency-lowering algorithms, tested every weekend.

They described the experience as "unnatural" and "noisy". A period of training, according to the previous studies, was expected to be helpful for listeners to get them accustomed to frequency-lowering. In this study, the listeners were requested to undergo training in the final five weeks, and the results of short sentence recognition taken every week were analyzed to track the effect of the training, as shown in Fig. 8. L5 chose M_1, so her first score was recorded after the first week. It was found that speech intelligibility kept increasing in the first two to three weeks, and then came to a stable value in the following three to four weeks. This indicated that with properly arranged training, it was possible for the users to become accustomed to the M algorithms in a period of six weeks.

3.6. Comparison of the two algorithms

As we supposed, by implementing different strategies for processing the original low-frequency content and by relocating the high-frequency content, the two proposed algorithms showed differences during parameter fitting, hearing and speech intelligibility tests, and users' preferences. The following is a brief summary of the findings:

The overlapped compression algorithm (M_1) did well in preserving the nature of the original low frequencies by allowing for mixing in a restricted region. If a patient was more concerned with hearing comfort and sound quality than with the other factors, M_1 was found to be an appropriate choice. The segmented compression algorithm (M_2) compressed both the low- and high-frequency region to keep them from overlapping. This would possibly cause decreased sound quality, but higher speech intelligibility was reported. This algorithm received greater preference in this study.

Performance in a noisy condition is another important hearing-aid algorithm. If the input sound contains noise, the noise will keep its level when compressed but will pile up when overlapped. The increase in the relative noise power may bring damage to the users' speech recognition because of their poor resistance to noise. In this aspect, M_2 was considered to have a better "anti-noise" quality than M_1, as indicated by the lesser decrease in the scores found in the noisy-condition tests.

4. Conclusions

Two objectives were achieved by this study. One was to use various parameter settings and frequency-lowering strategies to explain the frequency-lowering algorithms' performance in individual patients. Some rules were developed from the parameter fitting, and the corresponding features were found in the algorithm's effect on speech intelligibility and the listeners' subjective descriptions. Based on this, the causality of the algorithms, the parameters, and their practical effect were discussed. A comparison of both algorithms was conducted throughout this study.

The other target of this study was to evaluate the effect of frequency-lowering in Chinese speech recognition, especially for persons with severe hearing impairments in the high frequencies. Most previous studies have focused on the English and Japanese languages, so our investigation of Chinese phonemes, intonations, words, and sentence recognition can be seen as a preliminary feasibility investigation for the future applications of the frequency-lowering technique on Chinese speech processing.

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