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An Improved Multi-band Loudness Compensation Method Based on Nonlinear Frequency Compression for Digital Hearing Aids

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Abstract A comprehensive method applying a nonlinear frequency compression (FC) as complementary to multi-band loudness compensation is proposed, which is able to improve loudness compensation and simultaneously increase high-frequency speech intelligibility for digital hearing aids. First, to avoid the spectral distortions in the sounds, speech intelligibility-based frequency spectrum splitting is introduced. Then a nonlinear FC (NLFC) is also proposed to compress the high-frequency sounds to the lower bands where the audibility is available. Moreover, the introduced NLFC adjusts compression ratio (CR) based on the speech intelligibility percentage in different frequency ranges. Finally, an adaptive wide dynamic range compression (AWDRC) with a time-varying CR is applied to achieve adaptive loudness compensation, and prevent the stationary CR of the typical wide dynamic range compression (WDRC) from generating a negative impact on the speech. The experimental test results show that the mean speech identification is improved at least 20% points in comparison with the typical WDRC or the conventional FC.

Key words digital hearing aid; loudness compensation; AWDRC; frequency compression

一种改进的适用于数字助听器的基于非线性频率压缩的多通道响度补偿方法

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摘要 为了改进数字助听器中的响度补偿, 并提高高频部分的语音可懂度, 提出一种基于非线性频率压缩的多通道响度补偿的综合方法。首先, 为了避免语音的频率畸变, 基于语音可懂度进行频谱的多通道划分。然后, 采用一种非线性的频率压缩方法, 将高频部分的声音压缩至患者能听到的低频部分。所提出的非线性频率压缩方法是基于不同频段对语音理解度的贡献占比来改变频率压缩比。最后, 为了实现自适应的响度补偿同时防止传统宽动态范围压缩的固定压缩比降低语音质量, 采用一种随时间可变压缩比的自适应宽动态范围压缩方法。实验结果表明, 相对于传统的宽动态范围压缩和频率压缩方法, 该方法可以改善 20%的语音鉴别准确率。

关键词 数字助听器; 响度补偿; 自适应宽动态范围压缩(AWDRC); 频率压缩
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Hearing aid (HA) devices are currently the most popular methods for improving the communication ability of hearing-impaired (HI) individuals[1]. People with hearing loss usually suffer from both the loss of audibility and the reduction of hearing dynamic range. Generally they have smaller hearing loss in low frequency and greater in high frequency[2].

So far, wide dynamic range compression (WDRC) scheme is one of the most popular strategies adopted in HA to increase the loudness. WDRC uses static compression ratio (CR) to provide more amplification for the low-level sounds and less amplification for the high-level sounds[3]. While WDRC simultaneously fulfills the audibility and comfort requirements, it often reduces the performance of speech intelligibility[3–5]. Recently, Lai et al.[6] proposed a WDRC-based strategy-AWDRC, which dynamically adjusted CR according to the short-term dynamic range of the input signal. Experimental results indicated that AWDRC could provide better audibility than the conventional method that used a static CR.

However, WDRC or AWDRC strategy has a major disadvantage that this paper attempts to fix. In many cases, hearing sensitivity is very poor in the high-frequency band, where WDRC or AWDRC strategy just provides limited amplification. It is not feasible to sufficiently increase the gain to guarantee intelligibility, especially when the high frequency ranges have significant contributions to speech intelligibility[1]. Meanwhile, earlier investigation[7] indicated that the human listening awareness to speech does not depend on absolute frequency, but relative ratios.

For all these reasons, to improve audibility compensation and increase high-frequency speech intelligibility, simultaneously, we propose an improved method applying a nonlinear frequency compression technology as complementary to multi-band loudness compensation. The loudness compensation is based on AWDRC strategy. The frequency compression is applied to compress the high-frequency sounds to lower bands where audibility is available. And the introduced NLFC method improves the conventional methods[8] in the aspect that the frequency compression ratio is adjusted based on the speech intelligibility percentage in different frequency ranges. Furthermore, to avoid spectral distortions in the sounds[8], the frequency spectrum is split into six bands also based on the speech intelligibility percentage in different frequency ranges. Then for every band, the loudness compensation is processed, respectively. Based on the above improvements, the proposed method can fulfill the audibility and comfort requirements, as well as avoid the spectral distortions and achieve the preservation of intelligibility.

1 Typical WDRC

For better understanding, we briefly review the typical WDRC, and show its advantages and weakness.

A typical WDRC-processing input/output (I/O) function for a signal band is depicted in Fig. 1. HTn and HTu denote the normal and the patient's hearing threshold. DCTn and DCTu denote the normal and the patient's discomfort threshold. WDRC is used to compress the signal's dynamic range (HTn to DCTn) to fit the residual dynamic range (HTu to DCTu) of HI patient (e.g., in Fig. 1, HTn, DCTn, HTu and DCTu are 30, 120, 50, and 110 dB, respectively).

The signal processing is divided into four regions based on the input level.

1) A linear gain for input levels below the lower threshold (LTH).

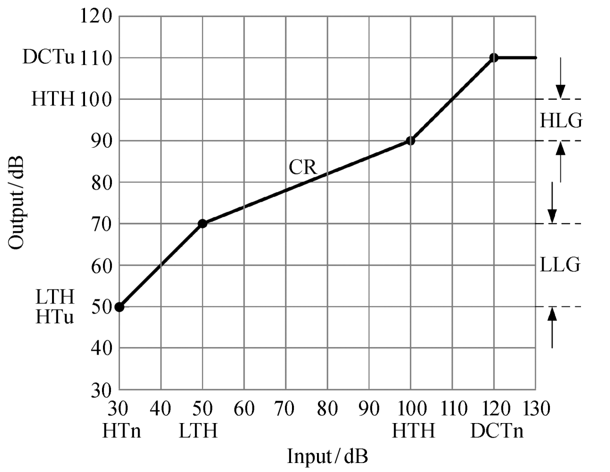


Fig. 1 Steady-state I/O function for a single frequency band in a digital hearing aid

2) WDRC for input levels between the LTH and the higher threshold (HTH), with the output level increasing by 1/CR dB for each dB increase in the input level (e.g., in Fig. 1, the output level increases by 2/5 dB for each 1 dB increase in the input level).

3) A linear gain for input levels above the HTH.

4) Compression limiting is activated for very high signal levels to keep the output lower than the predefined level (DCTu). A CR value of 1 equates to linear amplification for all except very high inputs. The shape of the I/O curve (Fig. 1) can be adjusted by varying low-level gain (LLG), LTH, high-level gain (HLG), HTH, and CR[4], where $\mathrm{CR} = (\mathrm{HTH} - \mathrm{LTH}) / [(\mathrm{HTH} + \mathrm{HLG}) - (\mathrm{LTH} + \mathrm{LLG})]$ (e.g., in Fig. 1, LTH, HTH, LLG and HLG are 50, 100, 20, and −10 dB, respectively).

Several studies have found that WDRC is more beneficial than linear amplification under speech-in-noise conditions. Olsen et al[9], Gatehouse et al. [10] and Shi et al. [11] all evaluated the benefits of the WDRC and linear amplification for speech audibility and intelligibility in different experimental methods. They found that speech audibility and intelligibility were higher for WDRC than for linear processing. Naylor et al. [5] measured the long-term signal-to-noise ratio (SNR) at the input and output of a hearing aid with amplitude compression by changing the CR under different input noise types, and band numbers. The results showed that the differences in these parameter values affected the output long-term SNR performance, with CR variations having the largest impact. However, the stationary CR of WDRC may have a negative impact on the long-term SNR perspective under some conditions. The output long-term SNR was constant for all values of input SNR when CR = 1:1, but it decreased as the CR increased and also after WDRC processing when the input SNR was positive, possibly because the gain of WDRC amplification increases low-level noise during speech pauses in the signal. During such pauses, the input level is dominated by the noise level, which therefore also determines the WDRC gain. Because the noise signal is lower than the speech signal for an SNR>0, the gain will be higher during these periods, and this reduces the output SNR relative to the input SNR. Based on the above assumption, when the CR of WDRC amplification is more than 1 and the input SNR is positive, the amplification method seems to have a negative impact on the long-term SNR perspective. Although the dynamic range of sounds can exceed 100 dB, the range required for daily oral communication is typically only about 40 dB. The range is even narrower if the dynamic range is measured for a short-term signal (e.g., 100 ms). Therefore, it is possible to use time-varying CR amplification while fulfilling the audibility and comfort requirements.

Even if the compression amplification applies the time-varying CR, the scheme just provides different amounts of amplification or compression for amplitude in different frequency regions. However, 1) in many cases, the hearing sensitivity is very poor in high frequency. It is not feasible to sufficiently increase the gain so as to achieve intelligibility. More important, the high frequency ranges have significant contributions to speech intelligibility[1]. 2) Meanwhile, earlier investigation[7] indicated that the human listening awareness to speech does not depend on absolute frequency, but relative ratios. So in order to improve audibility compensation and simultaneously increase high-frequency speech intelligibility, we propose an improved method applying a nonlinear frequency compression technology as complementary to multi-band loudness compensation. The following section would give more details.

2 Proposed Method

The system block diagram of the proposed method is shown in Fig. 2. First, to avoid spectral distortions in the sounds[8], the frequency spectrum is split into six bands based on the speech intelligibility percentage in different frequency ranges. Second, NLFC is introduced to compress the high-frequency sounds to lower bands where audibility is available. And the introduced NLFC improves the conventional methods[8] in the aspect that the frequency CR is

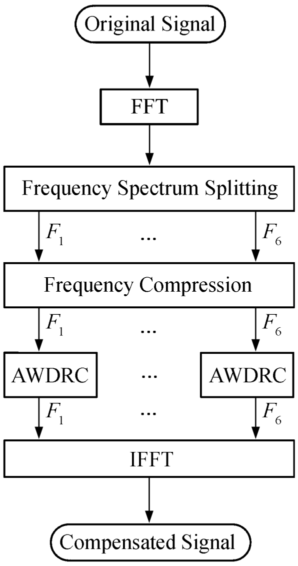


Fig. 2　System block diagram

adjusted based on the speech intelligibility percentage in different frequency ranges. Finally, the AWDRC with a time-varying CR is processed in every band to fulfill the audibility and comfort requirements. The proposed method is detailed in the following two subsections.

2.1　Frequency spectrum splitting and frequency compression based on speech intelligibility

As shown in Table 1, the different frequency ranges have different contributions to speech intelligibility[1].

So the frequency compression ratio is computed according to the relation in this study, which is more suitable to the person's perception ability. Furthermore, to avoid spectral distortions in the sounds[8], the frequency spectrum is split into six bands shown in Table 1. Then for every band, the compression amplification is processed, respectively.

Table 1　Relation between frequency range and intelligibility

Frequency range F_i /Hz	Intelligibility α_i/%
0–250	2
250–500	3
500–1000	35
1000–2000	35
2000–4000	13
4000–8000	12

According to the relation between frequency ranges and speech intelligibility in Table 1, the frequency range is divided into six bands according to octaves: 0–250, 250–500, 500–1000, 1000–2000, 2000–4000, 4000–8000 (in Hz). Every band is denoted as F_i ($i \in [1, 6]$) and its corresponding frequency range is $[f_{d,i}, f_{u,i}]$.

The nonlinear frequency compression method is adopted. Different from general methods, the frequency compression ratio was set according to speech intelligibility percentage in different frequency ranges. The basic principle is that the more the percentage is, the less the compression ratio is.

To fit the frequency compression scheme, a cut-off frequency (CF) is predefined individually for each hearing impaired individual based on the hearing aid fitting procedure. As a general rule, the cut-off was initially set to the lowest frequency at which conventional amplification provided the listener with inadequate audibility.

Generally, the CF (f_{CF}) is not less than 1 kHz, so the band F_1 and F_2 are always uncompressed. From Table 1, if the intelligibility percentage α_6 is supposed as 1, then α_5, α_4 and α_3 is approximately 1, 3 and 3, respectively. Assume that the f_{CF} is located in the band F_j whose frequency range is $[f_{d,j}, f_{u,j}]$ ($f_{d,j} < f_{\mathrm{CF}} \leqslant f_{u,j}$). Frequencies which are lower than $f_{d,j}$ is unchanged and those above this frequency are compressed and shifted to the remaining frequency range $[f_{d,j}, f_{\mathrm{CF}}]$. The frequency range which every band F_i is compressed to is $[\mathrm{fo}_{d,i}, \mathrm{fo}_{u,i}]$ denoted as FO_i. It can be mathematically described as follows:

$$\begin{cases} \mathrm{fo}_{d,i} = \mathrm{fo}_{u,i-1}, \\ \mathrm{fo}_{u,i} = \mathrm{fo}_{d,i} + \dfrac{\sum_{k=j}^{i}\alpha_k}{\sum_{k=j}^{6}\alpha_k}(f_{\mathrm{CF}} - f_{d,i}), \end{cases} \tag{1}$$

$i \in [j, 6]$.

Because two points determine a straight line, the linear relation between compressed frequency f_{out} and original frequency f_{in} is computed according to Eq. (2):

$$\frac{f_{\text{out}}-\text{fo}_{u,i}}{f_{\text{in}}-\text{fo}_{d,i}}=\frac{\text{fo}_{u,i}-f_{u,i}}{\text{fo}_{d,i}-f_{d,i}}. \tag{2}$$

Changing the form, f_{out} can be described as

$$f_{\text{out}}=\frac{\text{fo}_{u,i}-f_{u,i}}{\text{fo}_{d,i}-f_{d,i}}f_{\text{in}}+\frac{f_{u,i}\text{fo}_{d,i}-\text{fo}_{u,i}f_{d,i}}{\text{fo}_{d,i}-f_{d,i}},\ i\in[1,6]. \tag{3}$$

For example, assume that the CF is 3 kHz, so the f_{CF} is located in the band F_5 whose frequency range is [2000, 4000] and $j=5$. According to Eqs. (1) − (3), the relation of different frequency regions is shown in Table 2.

The relation of the frequency range between original signal and compressed signal (CF = 3 kHz) is shown in Fig. 3. From the graph, the frequencies below 2 kHz are uncompressed, and those above 2 kHz are compressed to regions between 2 and 3 kHz.

The introduced nonlinear frequency compression based on speech intelligibility compresses a frequency range above CF and shifts it to an adjacent lower-frequency area below CF to provide these compressed frequencies intelligibility. By adopting different frequency compression ratio, different frequency ranges are compressed, respectively. So its curve is segmented linear, and it can avoid the spectral distortions and achieve the preservation of intelligibility.

After the frequency spectrum splitting and frequency compression, we can process the AWDRC with the time-varying CR in every band to fulfill the audibility and comfort requirements.

2.2 AWDRC amplification scheme

To compress the signal's dynamic range to fit the residual dynamic range of HI people with the time-varying CR, each band introduces a feedback component whose input is the output of the band, and the output of the feedback component is used to adjust the CR of that band, as shown in Fig. 4.

Table 2 Relation between original and compressed regions

Original regions [$f_{d,i}$, $f_{u,i}$]	Compressed regions [$\text{fo}_{d,i}$, $\text{fo}_{u,i}$]
[0, 250]	[0, 250]
[250, 500]	[250, 500]
[500, 1000]	[500, 1000]
[1000, 2000]	[1000, 2000]
[2000, 4000]	[2000, 2500]
[4000, 8000]	[2500, 3000]

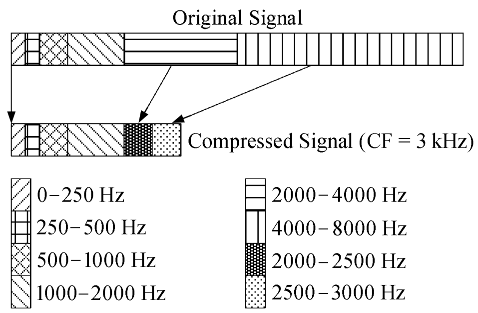


Fig. 3 Frequency range of original signal and compressed signal

The task of this feedback component is to perform a boundary check calculation, whereby the output level of the WDRC amplifier is estimated on a frame basis and the result is stored in a first-in-first-out queue of finite length L. When a new estimate is obtained, the maximum estimate (Maxest) and the minimum estimate (Minest) of the L estimates in the queue are used to drive the operations according to the AWDRC rules by the delta_gain parameter (with units of decibels per millisecond), which will increase or decrease the HLG or LLG in the interval between two consecutive estimates (i.e., delta_t), and

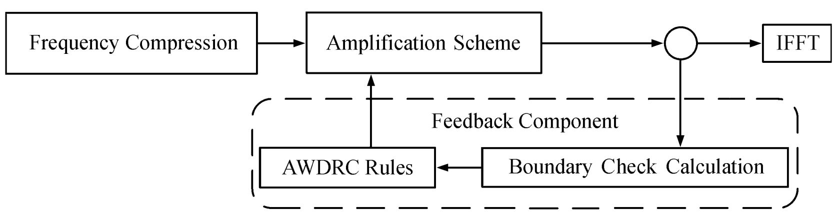


Fig. 4 Signal processing of a hearing aid with AWDRC amplification in one band

thus change the CR. The rules (which are executed every delta_t) governing the adaptive operations are described in detail below.

2.2.1 Decrease-CR Rule

The purpose of the Decrease-CR Rule is to keep the WDRC processing as close to linear as possible. As long as the Maxest is lower than the discomfort level (DCTu) and the Minest is higher than the hearing threshold (HTu), the LLG will be decreased and the HLG will be increased simultaneously by delta_gain, unless the CR already equals 1 (i.e., linearity). An example of the effect of this rule on the I/O function is shown in Fig. 5(a).

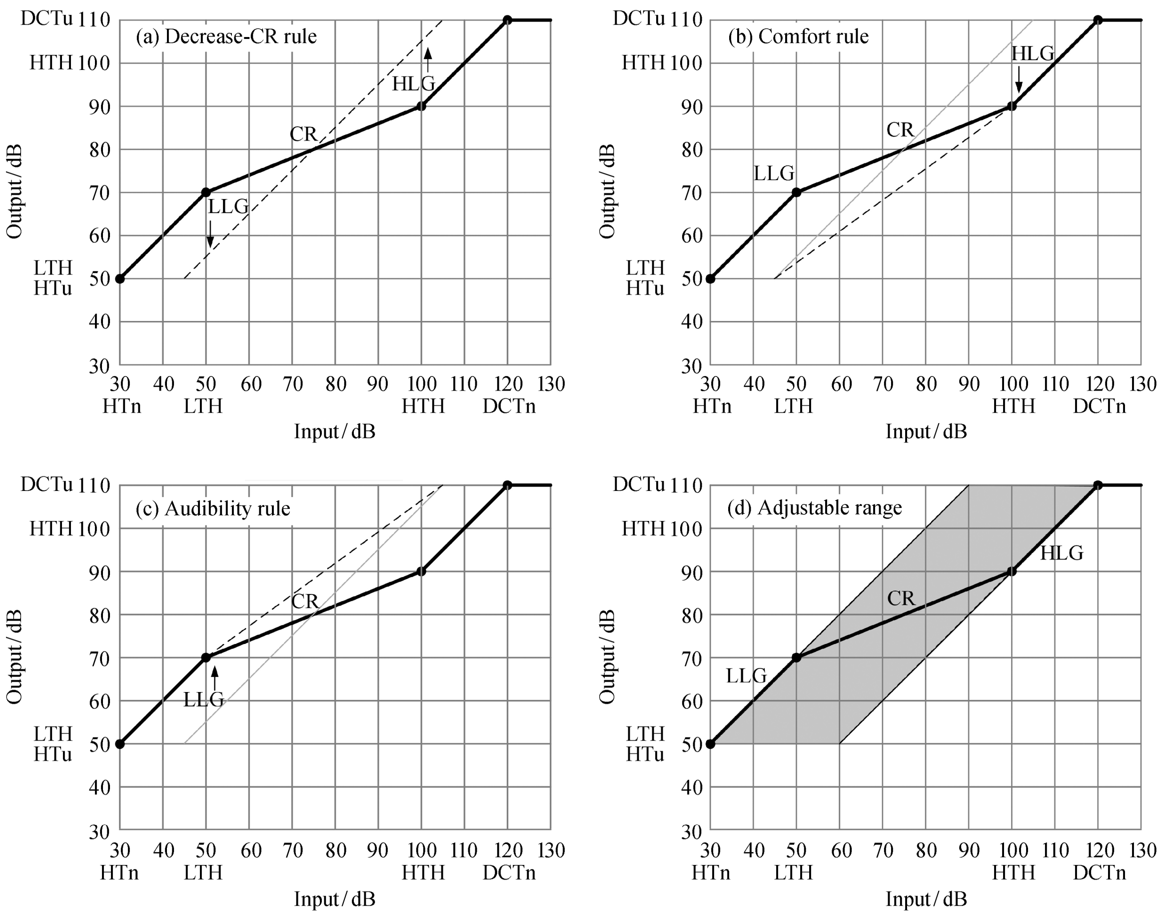
2.2.2 Comfort Rule

The purpose of the Comfort Rule is to ensure that the output sound will not be amplified to an uncomfortable level. When the Maxest is higher than the DCTu, the HLG will be decreased by delta_gain, unless the HLG is equal to the initially prescribed value. An example of the effect of this rule on the I/O function is shown in Fig. 5(b).

2.2.3 Audibility Rule

The purpose of the Audibility Rule is to ensure that the output sound is audible to the hearing-impaired individual. When the Minest is lower than the HTu, the LLG will be increased by delta_gain, unless the LLG is equal to the initially prescribed value. An example of the effect of this rule on the I/O function is shown in Fig. 5(c).

The priorities of these three rules decrease in the



The solid line is the original I/O function of WDRC, and the dashed line is the I/O function of AWDRC amplification adjusted adaptively for the decrease-CR rule (a), comfort rule (b), and audibility rule (c). The gray lines in (b) and (c) are for an I/O function adjusted by the decrease-CR rule. The shaded region in (d) represents the adjustable range of AWDRC amplification

Fig. 5 Example effects of AWDRC amplification on the I/O function. AWDRC amplification is based on the input sound modifying the I/O function adaptively

following order: Comfort Rule, Audibility Rule, and Decrease-CR Rule. In addition, the I/O relationship in AWDRC differs remarkably from its traditional static behavior in a single WDRC band. Fig. 5(d) shows an example of the I/O function of AWDRC, where the compressor seeks to match the design goals by using CRs that fall within the shaded area.

So AWDRC amplification will adaptively adjust the CR in an attempt to make all speech signals fall within the residual dynamic range of a hearing-loss individual, and prevent the stationary CR of WDRC from generating a negative impact on the long-term SNR perspective.

In summary, by means of splitting the frequency spectrum into six bands and the nonlinear frequency compression based on the speech intelligibility, the proposed method can avoid the spectral distortions and achieve the preservation of speech intelligibility. And in order to fulfill loudness compensation and comfort requirements, the proposed method introduces the time-varying CR multi-band compression amplification.

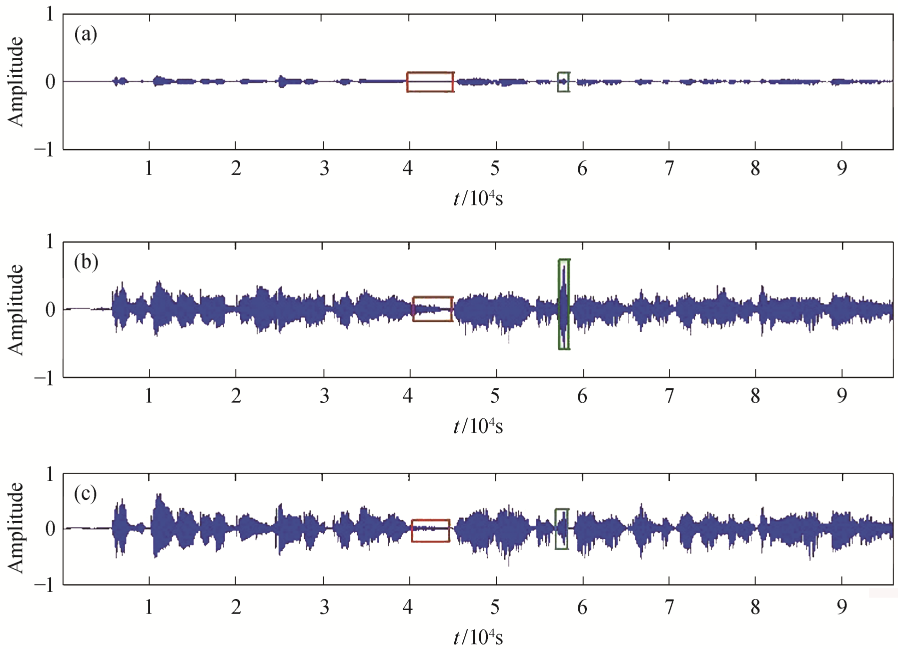
3 Experimental Results

In our experiment, the frame length is set as 20 ms, and the frame shift is half of frame length. The parameters in the NLFC and AWDRC is set according to the prediction from the fitting software. The delta_gain in AWDRC is set to 1 dB and delta_t is set to 100 ms, which corresponds to the LLG or HLG being increased or decreased by 1 dB every 100 ms. In addition, the attack and release times of AWDRC are set as 5 and 26 ms, respectively, which are the same values used by Naylor et al[5].

Fig. 6 illustrates the example waveforms of the original noisy speech, conventional WDRC processed, and the proposed method processed speech. The Fig. 6(a) shows the original noisy speech which includes normal speech and weak background noise. It can be seen from Fig. 6(b) that the conventional WDRC effectively increases the loudness comparing to the original noisy speech in Fig. 6(a). However, the conventional WDRC method fails to suppress the magnification of noisy signal (marked by red). What's worse, abnormal sound resulting from jittering of high-frequency signal is generated (marked by green) in Fig. 6(b). The situation may be owing to a larger gain for the low-intensity sounds than for the high-intensity sounds, and the noise level is generally lower than the speech level. Hence, the noise components are increased more by WDRC (which uses a static CR value). In contrast, the rules of AWDRC always try to decrease the CR value of WDRC in each band, so that the low-level noise during the pause segments are only amplified with a small gain. NLFC can reduce the jittering of high-frequency signals by compressing them to low frequency regions. These thereby improves the situation (Fig. 6(c)) relative to those from conventional WDRC (Fig. 6(b)).

Fig. 7 depicts the spectrograms of the original signal, the compressed signal with conventional FC method[8,12] and the processed signal with the proposed method (NLFC + AWDRC). The conventional FC method linearly shifts the frequencies between 0 and 8 kHz to the regions between 0 and 4 kHz, as shown in Fig. 7(b). The processed signal (CF = 4 kHz) with the proposed method is shown in Fig. 7(c), the signal with the frequencies below 2 kHz is uncompressed, and that above 2 kHz is compressed to regions between 2 and 4 kHz, as included in the red block. After the nonlinear frequency compression achieves the preservation of intelligibility, the loudness of the compressed signal is compensated by means of AWDRC to fulfill the audibility and comfort requirements, as shown in Fig. 7(c).

To test the subjective quality of the proposed method, we arrange 15 HI adults. They were experienced hearing aid users from Peking University Hospital. The parameters in NLFC and AWDRC are predicted individually from the fitting software. We respectively choose 10 words including the six international phonetic signs (IPA) /a/, /i/, /u/, /s/, /m/, and /sh/ form Implementation Summary of Mandarin Chinese Test[13]. Then the 60 words are respectively compensated with the typical WDRC, conventional



(a) Original noisy speech, (b) Compensated speech with conventional WDRC, (c) Compensated speech with proposed method

Fig. 6　Example waveforms of original noisy speech (a), conventional WDRC processed speech (b), and proposed method processed speech (c)

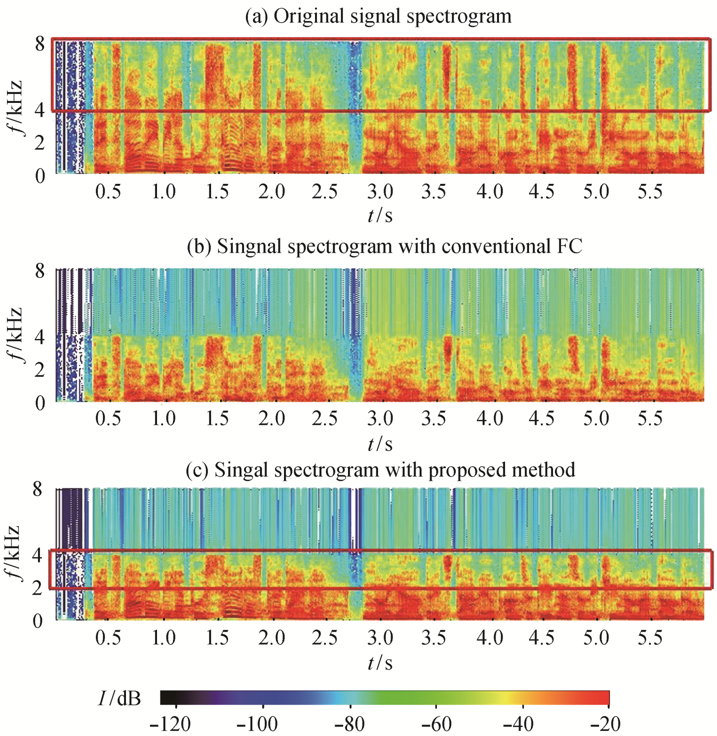


Fig. 7　Comparison of spectrogram

FC[8,12] and the proposed method (NLFC + AWDRC). The original and compensated words are recorded in the sound lab and played 5 times for the subjects. According to the subject's response, the proportion of words including IPA correctly identified was determined. The result is shown in Table 3.

Table 3　Correctly identified percentage for IPA

IPA	f /Hz	Correct percentage/%			
		Without aids	Typical WDRC	Conventional FC	Proposed
/m/	250–500	0	61.54	60.67	82.00
/i/	300	6.15	64.61	63.45	84.21
/u/	500	10.77	75.38	72.23	87.64
/a/	700	15.38	80.00	78.65	87.95
/sh/	2000–4000	0	45.23	56.38	80.65
/s/	3500–7000	0	40.00	52.00	78.54

From Table 3, we can tell that HI patients can barely recognize consonants without hearing aids. Meanwhile the recognition rate of voiced sound is fairly low, either. With the help of typical WDRC or conventional FC, most of voiced sound could be recognized by the patients. But in case of voiceless consonants, the improvement is not so significant, especially for /s/ and /sh/, whose dominant frequency is within the high frequency band. Obviously, the proposed method has the best performance, particularly for voiceless consonants which have large energy component in high frequency. The mean speech identification is improved at least 20% points in comparison with the typical WDRC or the conventional FC.

A limitation of this study is that the experiments involved software simulations only, without the use of any actual hearing aids, and did not consider the effects of various factors such as the characteristics of the microphone, receiver, and background noise conditions. Therefore, the overall effectiveness of the proposed method in real hearing aids needs to be investigated. Also, the objective and subjective speech intelligibility and sound quality should be evaluated in the clinical trials. These will be conducted in our future study.

4 Conclusion

This paper proposes an improved method applying a nonlinear frequency compression technology as complementary to multi-band loudness compensation, which is able to improve audibility compensation and simultaneously increase high-frequency speech intelligibility for digital hearing aids. The frequency spectrum is split into six bands based on the speech intelligibility to avoid the spectral distortions in the sounds. By introducing the nonlinear frequency compression whose CR is adjusted based on speech intelligibility percentage in different frequency ranges, the proposed method can achieve preservation of high-frequency speech intelligibility. With adding AWDRC scheme with the time-varying CR, the proposed method can make all speech signals fall within the residual dynamic range of HI individual with adaptive loudness compensation, and prevent the stationary CR of WDRC from generating a negative impact on the long-term SNR perspective. Simulation results indicate that the proposed method can fulfill the audibility and comfort requirements, as well as avoid the spectral distortions and achieve the preservation of intelligibility, especially for signal with abundant high frequency. The subjective test results show that the mean speech identification is improved at least 20% points in comparison with the typical WDRC or the conventional FC.

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