Loudness Compensation Method Based On Human Auditory For Digital Hearing Aids

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Abstract—**The intelligibility, the definition and the Comfort for digital hearing aids were reduced because of distorting formant of speech resulted by the existing loudness compensation method(LCM). In order to solve these problems, a LCM based on human auditory is proposed in this paper. This method adopts gammatone filter banks which can simulate the auditory model of human ear cochlea. Input signal is divided into 32 frequency bands by the filter banks. And then signal of each frequency band is compressed or amplified in accordance with the curve of hearing impaired. Experimental results show that the performance of the proposed method is better than the multi-channel LCM based on Bark filters.**

Keywords-digital hearing aids; multichannel loudness compensation; gammatone filter banks; human auditory;

I. INTRODUCTION

Hearing loss has become a world problem. Digital hearing aids is a powerful tool for most the hearing impaired. For the problem of the hearing impaired whose hearing threshold has been changed, a LCM was proposed. LMC is the most important core method in digital hearing aids. It can map normal earshot to hearing impaired earshot by amplifying or compressing the amplitude of speech signal according to the degree of hearing loss. As a result, it can help hearing impaired to improve sound clarity and comfort. So far, a lot of researches on LCM have been done by many scholars. At present, multi-channel loudness compensation methods have been achieved in digital hearing aids. For example, a loudness compensation method[1] based on 8-channel interpolating half-band filter with equal bandwidth was proposed by Thomas. A wavelet transform[2] was introduced by Li M. A loudness compensation method based on 16 channels nonuniform bandwidth filter banks[3] was proposed by Chong K. S. In these methods, formants of speech are distorted at the junction of filters. So speech intelligibility was reduced. To solve this problem, a multi-channel LMC based on formant[4] was introduced by Zhao Yi. However, in order to detect formant and redesign filter banks[5], the method has high computational complexity, so that it is rarely used to digital hearing aids.

The human ear's perception of sound frequency is nonlinear and approximate exponential relationship. So loudness compensation method based on uniform bandwidth filter banks is not satisfied auditory characteristics of the human ear. To solve this problem, multi-channel loudness compensation based on Bark filter banks[6~7] which is a non-uniform filter banks was proposed. In this method, the nonlinear perception of human ear is satisfied by Bark filter banks， but filters are independent to each other. If the compensation ratios of adjacent frequency band vary greatly, formants of speech signal will be destroyed easily in the filter at the junctions. For example, false formant was generated and formant was cut. For this reason, a LCM based on human auditory is proposed in this paper. In this method, gammatone filter banks which can simulate the auditory model of human ear cochlea were adopted. It can analog cochlea decomposition mechanism and active feedback of cochlea. In addition, the reasons of adopting gammatone filter banks are as follows. Firstly, the adjacent bands are smoothed because each filter has a long tail. It can protect formant from cutting. Secondly, for the most sensorineural hearing loss, cochlear membrance cells and retrocochlear neuropathy are the major cause of deafness. The flow chart of proposed method is shown in Fig1. Firstly, input speech signal is divided into 32 channels by gammatone filter banks. Secondly, input sound pressure level of each channel was calculated. Thirdly, output sound pressure level of each channel was calculated by the mapping curve of the input sound pressure level (SPLin) and the output sound pressure level(SPLout). Then, the compensation ratio was calculated by the input and output sound pressure level of each channel. Finally, the loudness compensation signals are obtained by superposing all output signals.

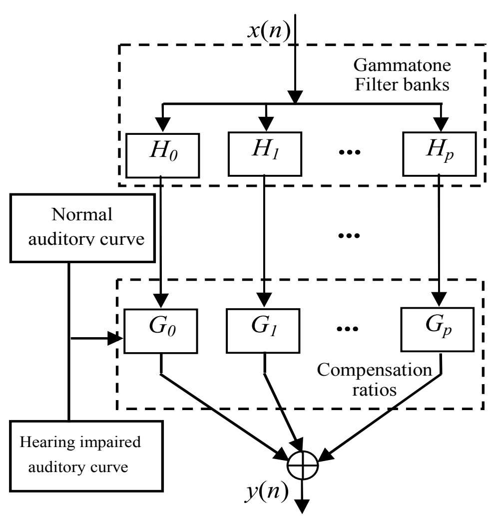

Figure 1. THE FLOW CHART OF THE PROPOSED METHOD

II. IMPLEMENTATION OF MULTI-CHANNEL LOUDNESS COMPENSATION

Human ear to perceive loudness of sound is a function of the frequency and SPL. There are different hearing thresholds, pain threshold and comfort threshold at different frequencies for human ear. Hearing threshold is the smallest sound pressure level (SPL), which can be just perceived by human ear. Comfort threshold is a SPL, which makes human ear can perceive more natural and more comfortable sound. Pain threshold is a SPL, which makes the human ear feel uncomfortable. The change of hearing threshold is the main factors of hearing loss.

According to the above analysis, the main purpose of loudness compensation method is to map normal earshot to hearing impaired earshot by amplifying or compressing the amplitude of speech signal. This method can improve sound clarity and comfort of hearing impaired. According to the human auditory characteristics, it is more reasonable that different channels have different compensation ratios based on the degree of hearing loss of hearing impaired. Based on the above analysis, a multi-channel loudness compensation method was proposed. Firstly, input speech signal is divided into n channels by filter banks. Then, different compensation ratios of different channels are calculated according to the curve of hearing impaired. Finally, the loudness compensation signals are obtained by superposing all output signals.

A. *Multi-channel filtering of speech signal*

Study of physiological showed that the cochlea and basement membrane is important organ in the human auditory system, which can perceive sound. The spectrum analysis of basement membrane is an important function to distinguish frequency and intensity information of voice.

In order to obtain the relationship between the frequency and location of the cochlea, many experiments have been done and equation (1) was obtained [8].

$$f = \alpha(10^{\beta(L-d)} - \delta) \tag{1}$$

In the equation (1), f denotes the frequency, L denotes length of the basement membrane (mm), d denotes the length between the top of basement membrane to frequency f . α , β and δ are equal to 165.4， 0.06， 0.88 respectively. L is equal to 35millimetre. Fig.2 showed the relationship between frequency and location of the cochlea. The frequency range is from 0 to 8000Hz. The length of the basement membrane at the top is equal to 0.

Fig.2 shows that the top of basement membrane correspond to the low-frequency perception region and the bottom of basement membrane correspond to the high-frequency perception region. And the relationship between the frequency and location of the cochlea approximate exponential distribution.

To simulate the mechanism of the cochlea, gammatone filter banks introduced by Johannesma P. I. M[9]. The filter banks simulate frequency decomposition mechanism of the human ear's basement. Impulse response function is shown in equation (2).

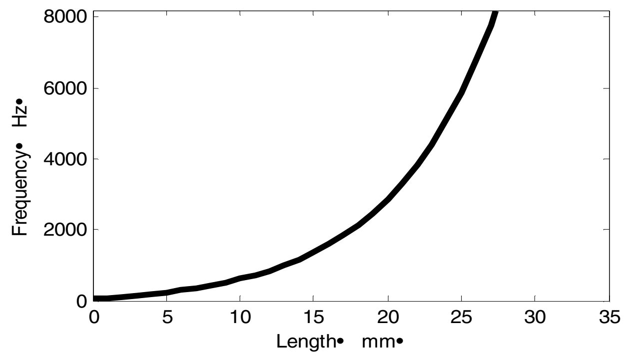

Figure 2. THE RELATIONSHIP BETWEEN THE FREQUENCY AND LOCATION OF THE COCHLEA

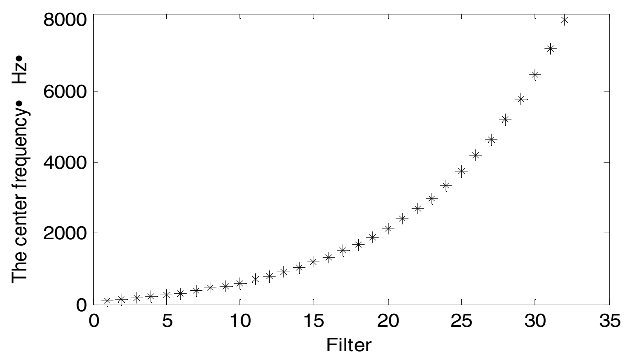

Figure 3. THE CENTER FREQUENCY OF GAMMATONE FILTER BANKS

$$g(f) = \begin{cases} t^{l-1}e^{-2\pi bt}\cos(2\pi ft) & if\,(t \ge 0) \\ 0 & else \end{cases} \tag{2}$$

In the equation (2), l denotes the order of the filter, which is equal to 4. b denotes the equivalent rectangular bandwidth. f denotes the center frequency of a filter. b and f satisfy Equation (3).

Fig.3 shows that the center frequency of the gammatone filter banks approximate exponential distribution. Compared with Fig.2, the center frequency of the gammatone filter banks well simulate the positions of frequency perception of basement membrane.

$$b(f) = c(24.7 + 0.108f) \tag{3}$$

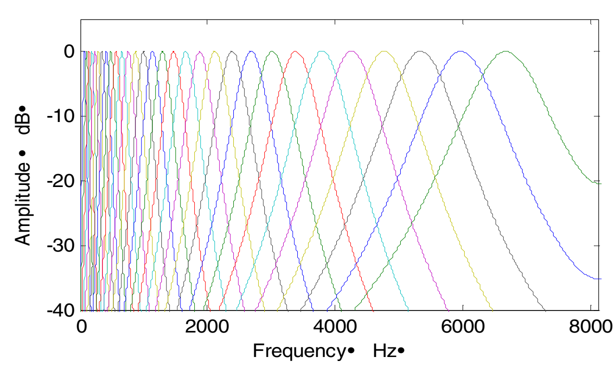

Figure 4. FREQUENCY RESPONSE OF GAMMATONE FILTERS

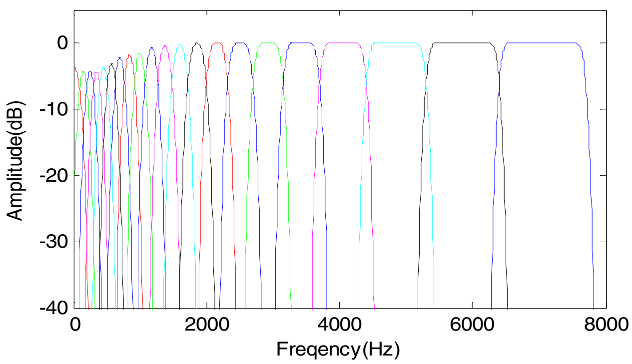

Figure 5. FREQUENCY RESPONSE OF BARK FILTERS

Fig.4 and fig.5 show the Frequency response of gammatone filter banks and Bark filters banks respectively. From these figures, the bandwidth increases with the center frequency. Compared with Bark filters, gammatone filters have long tail, which make filters overlap each other. Therefore, gammatone filters are not mutual independence. So the distortion of the formant structure at the junction of the speech signal can be well solved by gammatone filters.

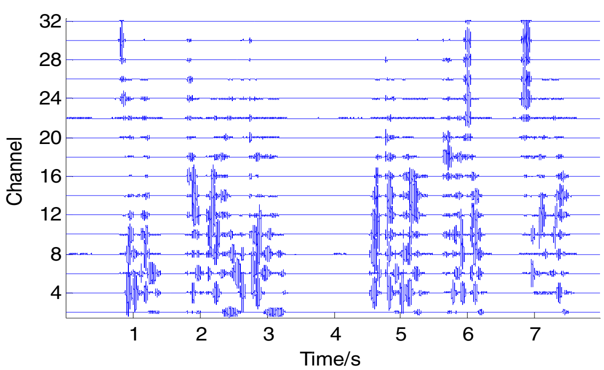

Figure 6. EACH SIGNAL FILTERED BY GAMMATONE FILTER BANKS

According to the human auditory characteristics, Gammatone filter banks were divided into 32 filters, Bark filter banks were divided into 22 filters. To compare distribution of signal filtered by nonuniform filter and uniform filter respectively, a uniform filter was designed, which were divided into 20 filters. A speech signal sampled at 16 kHz is filtered by gammatone filter banks and uniform filter banks respectively. Fig.6 shows that the signals filtered by gammatone filter banks (nonuniform filters) were distributed to all channels. Fig.7 shows that the signals filtered by uniform filter banks mainly concentrate on the top four channels. Because the bandwidth of a gammatone filter is narrow in the low frequency region and wide in the high frequency region, the frequency resolution is high in the low frequency and low in the high frequency. It is consistent with frequency resolution of the human ear. Therefore it is beneficial to loudness compensation for each channel. However loudness compensation signal filtered by uniform filter banks emphasis on the low-frequency region and the high-frequency signal is neglected, it is bad for hearing impaired of high-frequency loss.

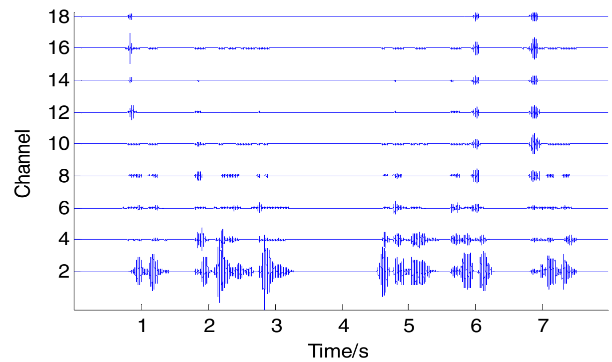

Figure 7. EACH SIGNAL FILTERED BY UNIFORM FILTER BANKS

B. Calculating channel compensation ratios

In the multi-channel loudness compensation method, SPLout of channel is calculated by formula (6). And then, the channel compensation ratios were obtained. Fig.7 shows the mapping curve of the SPLin and the SPLout. In the figure, *HTn*, *CTn*, *PTn* and *ESn* denote hearing threshold, comfort threshold, pain threshold and earshot of normal in a channel, respectively. *HTu*, CT*u*, *PTu* and *ESu* denote hearing threshold, comfort threshold, pain threshold and earshot of hearing impaired in a channel respectively. Mapping curve are composed of line *a* and *b*. Comfortable domain is at the junction of the two curves. Slope of line *a* and *b* satisfy the equation (4) and (5). Output sound pressure satisfies the equation (6).

$$Ra = \frac{CTu - HTu}{CTn - HTn} \tag{4}$$

$$Rb = \frac{PTu - CTu}{PTn - CTn} \tag{5}$$

$$SPLout = \begin{cases} 0 & ,0 < SPLin \le HTn \\ Ra(SPLin - HTn) + HTu & ,HTn < SPLin \le \mathrm{CTn} \\ Rb(SPLin - CTn) + CTu & ,CTn < SPLin \le PTu \\ PTu & ,SPLin > PTu \end{cases} \tag{6}$$

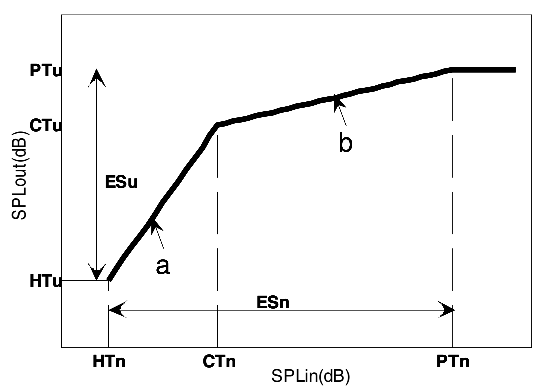

Figure 8. THE MAPPING CURVE OF SPLIN AND SPLOUT

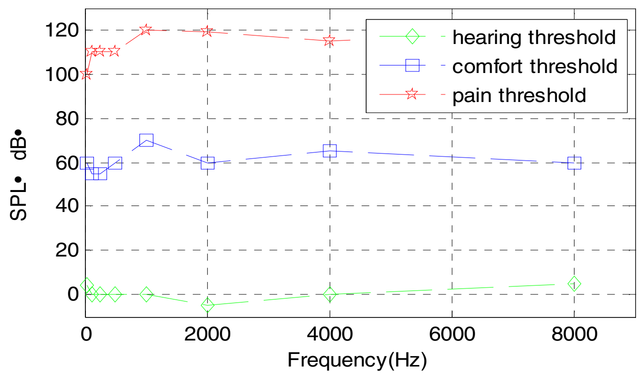

Figure 9. THE AUDITORY CURVE OF NORMAL

III. EXPERIMENTAL RESULTS AND ANALYSIS

In the experiments, speech signals sampled at 16 kHz and quantized with 16 bits were used for test. Speech signals come from NTT Standard voice library Chinese sub-libraries. And four kinds of auditory curve of hearing impaired (High-frequency loss, Flat, Descending, Ascending type) are shown in Fig.14, 15, 16 and 17 and the auditory curve of normal is shown in Fig.9.

Research shows that, in a speech synthesis, voiced was synthesized by the first three formants, which have good intelligibility. Therefore, formant of speech has great influence on intelligibility. In order to ensure the intelligibility of the speech, loudness compensation should as far as possible not to destroy the formant of speech. To analyze the formant structure of output speech signals, LPA spectral estimation of speech signals processed by proposed method and method of multi-channel loudness compensation based on Bark shown in Fig.10-13 are given.

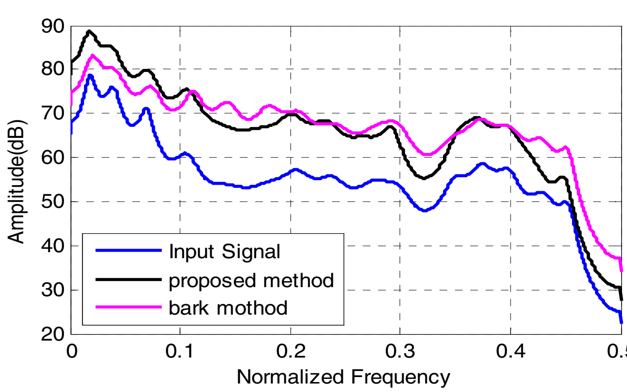

Figure 10. COMPARISON OF PROPOSED METHOD AND BARK METHOD(DESCENDING)

Fig.10-13 showed that spectral envelope and formant structure of speech signal processed by proposed method are similar to the original speech signal. However, speech signal processed by a method of a multi-channel loudness compensation based on Bark has greater distortion than original speech signal in positions of formant, which reduced speech intelligibility. Because each filter in Bark filter banks is independent, thus causing that false formant was generated and formant was cut at the junction of filters. However, each filter of gammatone filter banks has a long tail, which makes filters overlap each other. So the distortion of the formant structure at the junction of the speech signal can be well solved by gammatone filters. Compared with a method of a multi-channel loudness compensation based on Bark, the proposed method can improve the speech intelligibility and comfort for hearing impaired, because it can better protect formant than the Bark method.

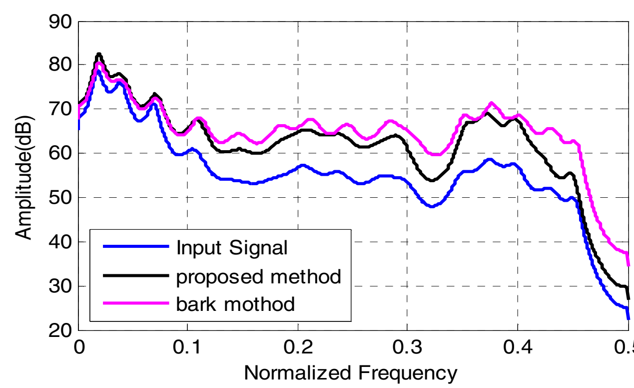

Figure 11. COMPARISON OF PROPOSED METHOD AND BARK METHOD (HIGH-FREQUENCY LOSS)

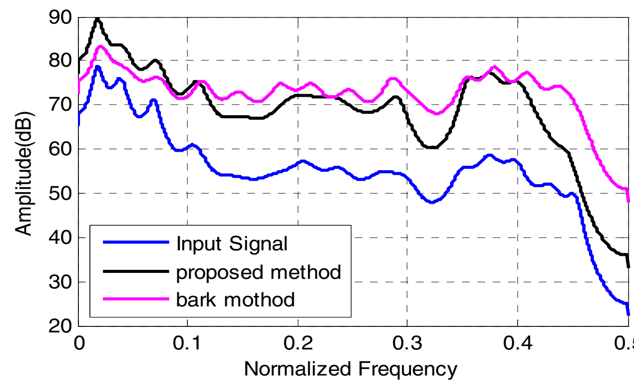

Figure 12. COMPARISON OF PROPOSED METHOD AND BARK METHOD(ASCENDING)

Fig.14-15 showed the SPLin and SPLout, hearing loss curve of ascending and descending. From the Fig.14-15, we can see that SPLin of each channel is less than hearing threshold. Because hearing threshold of hearing impaired is high in every channels, impaired ear cannot hear the original speech signal. Fig. 14-15 show that SPLout of Speech signal processed by proposed method is greater than hearing threshold of hearing impaired, hearing impaired can feel comfortable speech.

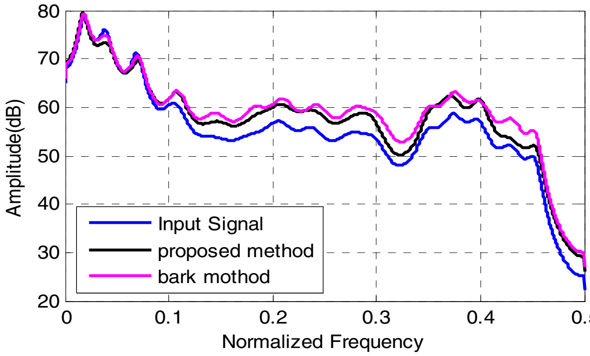

Figure 13. COMPARISON OF PROPOSED METHOD AND BARK METHOD(FLAT)

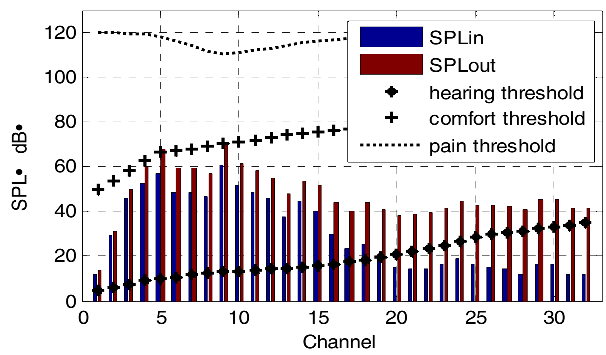

Figure 16. SPL OF INPUT AND OUTPUT SIGNAL(HIGH-FREQUENCY LOSS)

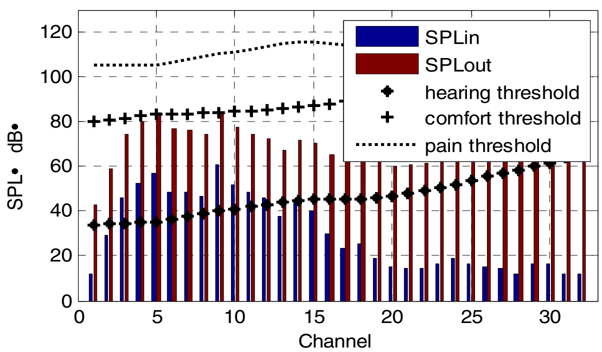

Figure 14. SPL OF INPUT AND OUTPUT SIGNAL(ASCENDING)

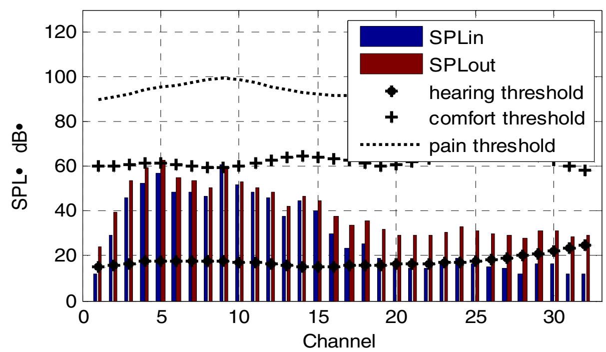

Figure 17. SPL OF INPUT AND OUTPUT SIGNAL(FLAT)

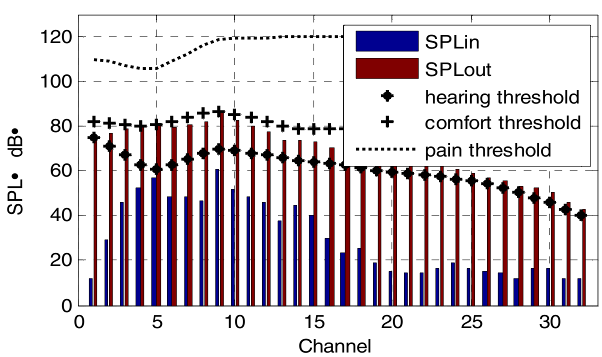

Figure 15. SPL OF INPUT AND OUTPUT SIGNAL(DESCENDING)

Fig.16-17 showed SPLin, SPLout, hearing loss curve of high-frequency loss and flat. From these figures, we can know when SPLin of each channel is greater than the threshold of hearing impaired; hearing impaired can only hear a small voice which can make the hearing impaired persons feel tired. When speech signal was processed by the proposed method, the magnitude of the signal of each channel is increased so that the intelligibility and comfort were improved for hearing impaired.

IV. RESULTS OF CLINICAL TRIALS

In order to verify the performance of the proposed method, subjective test is used. Test objects are several hearing impaired whose hearing loss are mild, moderate and severe separately. Test equipment shown in Fig.18 are pure tone eudiometry equipment and anechoic chamber manufactured by the US Starkey company, provided by Beijing China Hongsheng technology Co., Ltd. Materials were 96 sentences speech signals of 4 males and 4 females, which come from NTT standard speech library Chinese sub-libraries. According to the test requirements, the sound pressure levels of the 96 sentences were adjusted to 20db, 30db, 40db, 50db, 60db and 70db, and the 576 sentences speech signals as the test speech signals. Then, based on different hearing curves of hearing impaired, original sentences were processed by the proposed method and a method of multi-channel loudness compensation based on Bark respectively. Hearing impaired heard original sentences and loudness compensation signals which processed by the proposed method and the Bark method respectively, and the test results were recorded. The result is shown in table I.

As shown in the Table I, the proposed method is more effective for hearing impaired to improve the intelligibility than the Bark method. Testers generally felt that the speech signals proposed by the Bark method have “Zi Zi” noise and thought the auditory comfort of the speeches processed by the proposed method is better than Bark method. The reason for this result is that each filter of Bark filter banks is mutually independent. If compensation ratio of adjacent channels varies greatly, it could destroy speech formant at channel junction, such as formant shift, disappeared and even false formant. Compared with gammatone filter banks, each filter of gammatone filter banks has a long tail, which can make filters overlap each other. As the result, the distortion of the formant structure at the junction of the speech signal can be well solved.

Figure 18. TEST ENVIRONMENT

TABLE I. THE RESULT OF SUBJECTIVE TEST

SPL（dB）	**Correct rate（%）**		
	Original speech	***Proposed method***	***Bark method***
20dB	0%	30%	32%
30dB	10%	55%	45%
40dB	35%	62%	50%
50dB	40%	67%	61%
60dB	70%	82%	70%
70dB	75%	88%	77%

V. CONCLUSION

In the paper, a multi-channel LCM based on human auditory characteristics is proposed. The method used a gammatone filter banks which can simulate the auditory model of human ear cochlea. The proposed method and the multi-channel loudness compensation based on Bark are compared by analyzing from two aspects of subjective (clinical trials) and objective (LPA spectral), the proposed method can better protect formant of speech than the Bark method, the clinical trial results show that the speech intelligibility and comfort are improved for four kinds of typical hearing loss such as Descending, Ascending, Flat and High-frequency loss type. Experimental results show that the proposed method surpass a multi-channel loudness compensation based on Bark. Therefore, the proposed method has a good reference value for the loudness compensation design of digital hearing aids and practical application.

ACKNOWLEDGMENT

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