디지탈 보청기의 이득보상기법에 대한 연구

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Dynamic Loudness Compensation for Digital Hearing Aids

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ABSTRACT

This paper presents a new method which compensates loss of loudness for digital hearing aids. Loudness grows more rapidly in frequency domain with substantial shifts of hearing threshold, so that loud sounds reach the uncomfortable sound level (UCL) at about the same physical stimulus level as with normal hearing. The result is a compression of the available dynamic range of hearing. Many techniques have been developed to compensate for hearing losses. In this paper, we propose a digital hearing aid which uses a single digital filter for reducing distortion and the fuzzy function to calculate gain factors. This function describes how much gain is needed for every frequency to restore loudness perception of a normal ear.

INTRODUCTION

An Impaired person with sensorineural hearing loss suffers not only from a reduced ability to hear low intensity sounds, but also from difficulties in the discrimination of sounds which is above threshold of hearing level. The most common complaints of patients with sensorineural hearing loss are of difficulty in understanding speech in a noisy environment and their impaired mapping between the sound pressure level (SPL) of a natural acoustic sound and the perceived loudness of this sound.

Conventional hearing aids does not satisfy the needs of impaired persons who suffer from sensorineural hearing losses. The focus is on the sensorineural hearing losses, especially according to recruitment of loudness.

Multiband compensation algorithms were implemented which operates two, three, even twenty-four band filter channels. Unfortunately, multiband compression has several problems of its own. First, its non-linearity is producing harmonic and intermodulation distortion. Second, multiband compression produces a smoothing or flattening of the sound spectrum. This may make it more difficult for the listener to detect the salient features of the spectrum. Finally, the system has many parameters such as filter and compression characteristics for each channel, attack and release time. From there, system is large. The aim of our study was the real-time implementation of an algorithm which is consisted of a single digital filter for compensating loss of loudness with less distortion and modification of the short-time amplitude spectrum. A real-time digital hearing aid system with a general-purpose digital signal processor (DSP) board is implemented in the personal computer.

DYNAMIC RANGE OF HEARING

A Hearing impaired person with loudness recruitment has a reduced dynamic range hearing threshold and uncomfortable level. Figure 1 shows the approximations of equal loudness contours of a normal and impaired ear. At high signal intensities, loudness is about the same for normal and impaired persons. However, at lower signal levels, loudness is a function of hearing threshold, which is different for normal and impaired persons, and the UCL is about the same. This shows the reduction of the dynamic range of hearing.

The relation between loudness, which is a subject measure, and intensity, which is physical measure of signal energy, can be approximated as follows:

$$L=\left(I/I_O-I_{th}/I_O\right)^{\alpha} \tag{1}$$

L is loudness in sones, I_O corresponds to a reference sound intensity, and I_{th} is the sound intensity at the hearing threshold. The reduction ratio (rr) of the dynamic range of an impaired person to that of a normal person at a frequency (fr) can be represented by

$$rr=\frac{\log UCL-\log TH_N(fr)}{\log UCL-\log TH_I(fr)} \tag{2}$$

where UCL is uncomfortable hearing level, TH_N (fr) is threshold of a normal person at the frequency (fr), and TH_I (fr) is that of an impaired person.

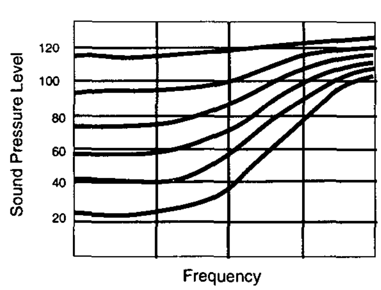

Figure.1 Dynamic range of impaired hearing

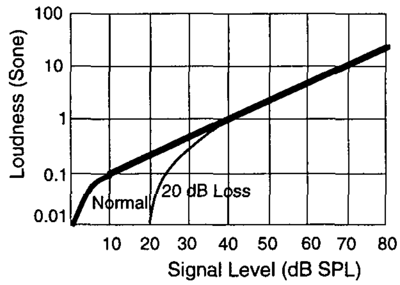

Figure.2 Comparison of normal and impaired loudness relations

A loudness compensation function is needed to restore the reduction of the dynamic range of an impaired person. The aim of study was the practical and real-time implementation of an algorithm which compensates reduced dynamic range for sensorineural hearing losses.

METHOD

The digital loudness compression hearing aid system consists of a general purposed DSP board (Ariel's DSP56001) for a personal computer with analog front ends (16 bit analog-to-digital converter and a digital-to-analog converter). The system is similar to the digital master hearing aid by Levitt[1]. Figure.3 shows a block diagram of the proposed hearing aid system. The sound signal is sampled at a rate of 10 k Hz, segmented into short time blocks (12.8m second). The frequency and spectrum of the input signal are calculated by using short-time FFT and the average intensities are estimated by loudness estimation block at 6 bands (below 250, 250 Hz to 500 Hz, 500 Hz to 1k Hz, 1k Hz to 2k Hz, 2k Hz to 4k Hz, and over 4k Hz). The modified spectrum is calculated by multiplying each intensity value with a gain factor that is determined by a membership function of the gain estimation at each block. This function describes how much gain is needed for each band, and is considered for each impaired person and situation. To prevent a spectrum flattening through independent modification, an interpolation is used over the whole frequency band. This modified spectrum is then transformed back into time domain, and the signal is sent to a reconstruction filter and a digital-to-analog converter.

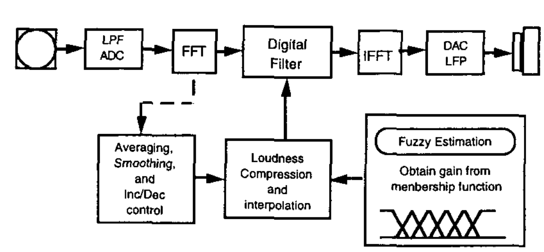

Figure.3 A Block diagram of the proposed digital hearing aid

DISCUSSION

A digital hearing aid faces new era of progress in the field of hearing aids. Conventional hearing aids have limited benefit to the many kinds of hearing losses, especially sensorineural hearing losses. A digital hearing aid can compensate sensorineural hearing losses. Many papers and researchers concerning this hearing loss suggest following summary of the deficits, such as loss of threshold sensitivity, compression of dynamic range, and loss of sharp tuning or auditory acuity. The tuning curves of auditory nerve fibers are widened in the damaged ear and some nonlinear processes, like suppression, are modified. This damage leads to increased masking and decreased ability to separate sounds of different frequencies; these deficits are particularly noticeable in difficult listening environments, especially those involving background noise. The current generation of digital hearing aids address the first two deficits and succeed reasonably well in overcoming them. Our future work will be focus to achieve a significant increase in hearing aid performance by addressing the third deficit.

CONCLUSION

The proposed evaluation system for practical digital hearing aids shows many benefits compared to conventional and programmable hearing aids. The aid can be adjusted to suit the individual patient, so as to deal with a wide range of degree and pattern of hearing loss. This eliminates many of the difficulties associated with the hearing aids fitting. We need more practical research and a wearable digital hearing aid. The new digital hearing aids algorithm will have a function to compensate loss of sharp tuning and auditory acuity.

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