감음신경성 난청의 모델링을 통한 라우드니스 누가현상의 시뮬레이션

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Simulation of the Loudness Recruitment using Sensorineural Hearing Impairment Modeling

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ABSTRACT

With the advent of high speed digital signal processing chips, new digital techniques have been introduced to the hearing instrument. This advanced hearing instrument circuitry has led to the need for and the development of new fitting approach. A number of different fitting approaches have been developed over the past few years, yet there has been little agreement on which approach is the "best" or most appropriate to use. However, when we develop not only new hearing aid, but also its fitting method, the intensive subject-based clinical tests are necessarily accompanied. In this paper, we present an objective method to evaluate and predict the performance of hearing aids without the help of such subject-based tests. In the hearing impairment simulation (HIS) algorithm, a sensorineural hearing impairment model is established from auditory test data of the impaired subject being simulated. Also, in the hearing impairment simulation system the abnormal loudness relationships created by recruitment was transposed to the normal dynamic span of hearing. The nonlinear behavior of the loudness recruitment is defined using hearing loss functions generated from the measurements. The recruitment simulation is validated by an experiment with two impaired listeners, who compared processed speech in the normal ear with unprocessed speech in the impaired ear. To assess the performance, the HIS algorithm was implemented in real-time using a floating-point DSP.

INTRODUCTION

With the loss of hearing, a person is restricted from his or her normal social activity. Recent developments in digital technology have offered new possibilities of noticeable advances of hearing aids. Using digital technology, it is possible to equip hearing aids with powerful features, such as nonlinear amplification, noise reduction, and enhanced fitting algorithms, these are often difficult to implement with analog circuits [1,2].

Although various ideas and methods have been suggested so far, there still exist many problems to be solved to meet the urgent requirements of the hearing impaired. In addition, evaluation of new ideas in hearing aids is necessarily accompanied by intensive clinical tests. However, subject-based clinical tests require much time and cost. Moreover, their potential problems are that 1) it is difficult for a patient to be involved continuously throughout the entire session, and 2) unreliable response from too old or too young aged patients. Rutledge [3] modeled impaired listener's transfer function using neural network and developed an objective measure from the function. And Chabries's approach [4] involved the modeling of an impaired listener's parameters based on a filterbank system when developing hearing aid. Rutledge's neural network is hard to manipulate auditory parameters, and Chabries's filterbank method is too complex to implement in real-time system.

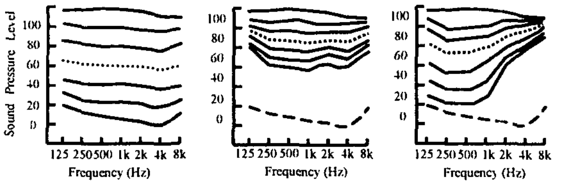
The audible thresholds of the normal listener are much lower than the range where most of speech signal components exist. However, as a hearing loss process proceeds, the audible thresholds begin to rise, so that it becomes increasingly difficult for the impaired person to hear weak level sounds. In the conductive hearing loss caused by problems in the outer ear and/or the middle ear, the sound level is reduced uniformly throughout the audible frequency region. Thus, in terms of the hearing impairment modeling, this type can be feasibly modeled with linear electronic circuits. But, for sensorineural hearing loss, the problem is more complex because it is caused by problems in the cochlea or in the auditory nervous system. In this case, nonlinear modification of the audible range can be occurred a phenomenon known as *loudness recruitment* [5]. Thus, sensorineural hearing impairment is mainly characterized by two parameters: elevation of the hearing threshold, and nonlinear mapping between the sound pressure level of natural acoustical signals and the perceived loudness.

We present a simulation tool for the sensorineural hearing loss and the effect of loudness recruitment. In our hearing impairment simulation (HIS) tool, the impairment is modeled using the data from auditory tests conducted for an impaired patient with sensorineural hearing loss. The key feature of the HIS tool is its ability to simulate the feeling experienced by the impaired listener and make it possible to perform subjective tests without hearing impaired patients. Its best application would be in the evaluation of the hearing aids or algorithms for hearing aids. To test the effectiveness of the proposed HIS tool, it is implemented in a real-time system employing a floating-point digital signal processor (Motorola DSP96002). The processed signals are presented to normal subjects and their responses are then investigated. The test results of normal listeners are compared with those of impaired subjects.

MATERIALS AND METHOD

1. The Hearing Impairment Simulation (HIS)

Hearing loss causes degradation in loudness perception, which can be modeled by matching the output of the normal ear to that of the impaired ear. By focusing on the



(a) Normal (b) Conductive Loss (c) Sensorineural Loss

Fig.1 Types of Hearing Losses

auditory function rather than on the characteristics of the input audio signal, the resultant signal processing will be effective independent of the auditory stimulus. However, the structure of the modeling system will limit the complexity of the algorithm that can be implemented in a hearing aid system. The modeling process is based on a system operating on the full frequency bands and dynamic range. A nonlinear dynamic processing which transforms input according to the level and frequency of the input is required.

In order to have an accurate model for the impaired mapping associated with the loudness recruitment, the critical band processing [6] is employed. To estimate the energy of the input signal at each critical band, the input signal sampled at 16kHz is segmented with Blackman window. The 16kHz sampling rate covers 20 critical bands. Each segment is 128 samples corresponding to 8 ms time period. Successive short-term spectra are then calculated using a 128-point FFT. In the algorithm, the short-term energy rather than the magnitude is computed in each critical band to reflect psychoacoustical features.

To define the impaired mapping between the sound pressure level of the natural acoustical signals and the perceived loudness of the impaired auditory system, hearing loss functions (HLFs) are generated from the measurements. These functions describe how much gain is needed for every frequency band and the level to restore normal loudness perception. The HLF's are generated from three auditory parameters: threshold of hearing (TH), most comfortable level (MCL), and uncomfortable level (UCL) which represents the upper limit of hearing range.

Simulating a phenomenon of loudness recruitment modeling may require all the information on equal loudness curves of the impaired and the normal subject's. However, it is impractical to measure the loudness for every input level at every frequency. Furthermore, loudness itself is based on subjective feelings, so it is hard to obtain accurate data solely from the measurements. Fortunately, it is known that the MCL represents the characteristics of equal loudness curves and that it can be measured relatively

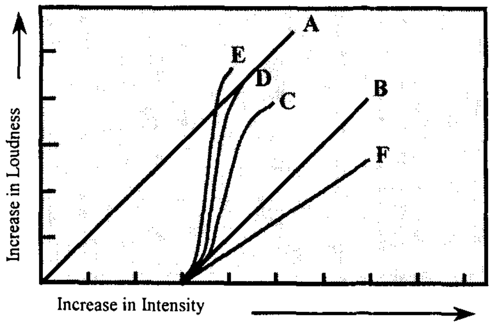


Fig.1 The growth of loudness with increasing intensity
A: Normal, B: No Recruitment, C: Incomplete Recruitment, D: Complete Recruitment, E: Hyper-recruitment, F: Decruitment

precisely, especially in most cases of sensorineural-type hearing loss [4]. From this point of view, the MCL measurements are used as a reference parameter in loudness level that prescribes a proper mapping from audible range of the normal listener's into that of the impaired listener's.

Based on these aspects, the energy estimates in the 20 critical bands are applied to HLFs and the hearing loss gains are computed. However, in the implementation stage, we computed the gains only at five center frequencies of octave bands, i.e., 250 Hz, 500 Hz, 1 kHz, 2 kHz, 4 kHz, and then interpolated them linearly while all spectral components in a critical band have the same gain.

To produce outputs modified according to the frequency gains obtained from the HLF, a frequency sampling filter is designed. The filter obtains coefficients from the magnitude response function sampled uniformly in the frequency domain, it provides an effective and flexible method to implement loudness attenuation [7]. Furthermore, this approach enables one to avoid the problems associated with the filter-bank approach. The filter coefficients are updated at every input block and the input signal is finally modified via convolution operations.

2. Experimental System Setup

The schematic diagram of the system setup for the experiments is shown in Fig. 4. A floating-point digital signal processor (33 MHz Motorola DSP96002) was used to implement the algorithm in real-time. The processor is a part of an Ariel DSP board installed in a PC-bus slot with a 16-bit AD/DA CODEC unit.

The Grason-Stadler's SGI 61 audiometer was used to supply the input signal to the system. The HIS system was calibrated to have 0 dB input-output gain, so that the

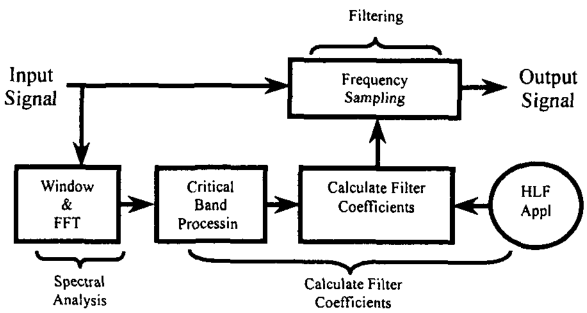


Fig.2 Block diagram of HIS System

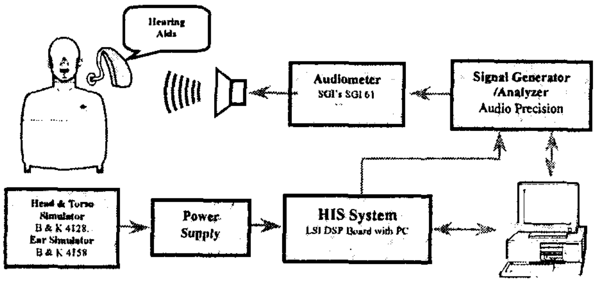
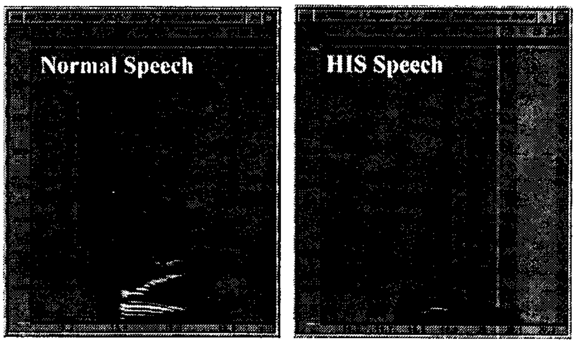


Fig.4 Test system set-up



(a) (b)

Fig.5 An Example of Normal speech (so:l) and processed speech by HIS

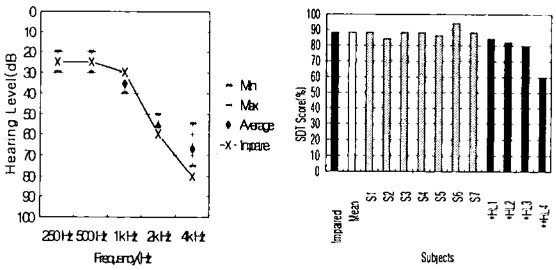
attenuation by the system could be directly observed at the output of the system. The total processing time of the HIS algorithm was about 52% of the maximum processing time in the real-time DSP board.

RESULTS

Two kinds of tests were conducted to evaluate the performance of the HIS system. The first test was to evaluate the numerical accuracy of the system. The HIS system and the audiometer were set to generate the same level of test signals and the output levels were measured for comparison. The second test was intended to simulate the loudness sensorineural hearing impairment. The audiological data of the impaired subject was given to the HIS system a priori, and the processed output was presented to normal subjects to measure their responses. The pure tone audiometry and the speech discrimination test (SDT) were performed to compare the HIS test results with the audiometry data of hearing impaired listeners. Finally, we evaluated three digital hearing aid algorithms using the HIS system and measured the SDT scores of them.

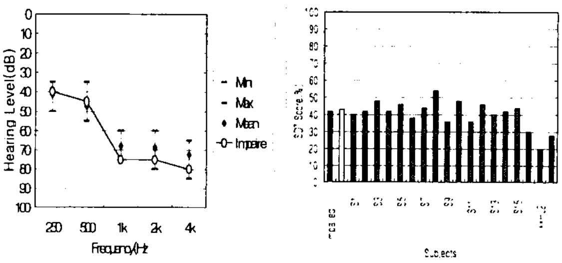
1. Objective Measurement of HIS Output

To assess the numerical accuracy of the HIS system, the conductive hearing loss i.e., a uniform hearing loss over the entire frequency was assumed. A 90 dB pure tone was



(a) (b)

Fig.6 Hearing thresholds(a) and SDT scores(b) of hearing impairment model and HIS-processed listeners. Data(a) represent pure tone audiogram of the model and average, minimal(Min), maximum(Max) values from 7 HIS-processed subjects. In graph (b), subjects are composed of 7 normal listeners(S1-S7) and 3 impaired listeners with C5 dip(*HL 1-3 : 30-45 dB HL at 4 kHz), and a high tone loss(**HL4 : 10 dB HL at 4 kHz, 55 dB HL at 8 kHz).



(a) (b)

Fig.7 Hearing thresholds(a) and SDT scores(b) of hearing impairment model and HIS-processed listeners. Data(a) represent pure tone audiogram of the model and average, minimal(Min), maximum(Max) values from 15 HIS-processed normal subjects. In graph(b), subjects are composed of 15 normal listeners(S1-S15) and 3 impaired listeners with C5 dip (*HL 1 : 40 dB HL at 4 kHz), and two high tone losses(**HL 2-3 : 10 dB HL at 4 kHz, 45, 30 dB HL at 8 kHz). SDT: speech discrimination test, HIS: hearing impairment simulation

applied to the input and results were obtained as RMS values measured at 5 frequencies; 250 Hz, 500 Hz, 1 kHz, 2 kHz, and 4 kHz. The measured errors were smaller than 1.6 dB. Furthermore, the errors were considered insignificant when compared with the measurement error that may occurred during the subjective tests.

2. Simulation of the effect of loudness recruitment

In the second test, two sensorineural hearing impairments were modeled using the HIS system. We chose high frequency hearing impairments to simulate. The first hearing model was the hearing of 64-year old male subject with the sensorineural hearing impairment. The target subject had a severe degree of high frequency hearing loss; 78 dB HL of the MCL and 88% of the SDT score. The HIS system was tested by 11 hearing listeners, aged between 21 to 33. The hearing thresholds of the 7 listeners were qualified as normal and the other four subjects had high frequency hearing loss. Three subjects showed the C5 dip and the other showed high frequency hearing loss above 8 kHz. The other simulation hearing model was the hearing of 40-year old male subject. The subject had a similar degree of high frequency hearing loss; 74 dB HL of the MCL but he had only 42% of the SDT score. The HIS system was tested by 18 hearing listeners. The hearing thresholds of the 15 listeners were also qualified as normal and the other three subjects had high frequency hearing loss.

The thresholds of hearing were measured with the HIS system. The results are presented in Fig. 6 (a) and Fig. 7 (a). As shown in the figures, the threshold levels of the tested subjects are different from those of the case without the HIS system. Although these results varied from person by person, the averaged thresholds exhibited similarity among the target model and the simulated ones.

The SDT scores were measured along with the threshold tests. The results are shown in Fig. 6 (b) and Fig. 7 (b). The bars with the label of 'Impaired' and 'Mean' indicate an SDT score of our subjects. The SDT scores again show that there exists a close resemblance among the target model and the simulated ones. Most of the subjects showed the SDT scores comparable to the discrimination score of the target impaired listener. From the results shown in Fig. 6 and 7, it could be said that the present HIS system was successful in simulating sensorineural hearing impairment using the normal listeners.

CONCLUSION

A hearing impairment simulation tool was developed and its performance was evaluated using a real-time system. To modify the natural input sound into an impaired one, a frequency sampling filter was designed, whose frequency response is continuously adjusted according to hearing loss functions. Because the selection of the gains for the FIR filter design was controlled by the energy values in 20 critical bands, the signal is nonlinearly processed as a function of its frequency and intensity. The experimental results of the HIS system showed that the HIS system implemented the hearing loss model with a tolerate margin of error. Furthermore, subjective tests conducted with normal subjects confirmed the effectiveness of the HIS system in simulating the effect of loudness recruitment for the sensorineural hearing impairment, which was clearly indicated in the measurement results of both the thresholds of hearing and the SDT.

So far, we have shown that the HIS system developed in this study can simulate sensorineural hearing impairment and it is quite worthy in using the system for providing the feeling of the hearing impairment to normal subjects. Moreover, the system can be used for more practical issues such as evaluating conventional hearing aids and developing new fitting methods for the digital hearing aids. More work regarding these issues is currently under way.

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