감음신경성 난청의 청각 모델링을 통한 보청기 알고리즘 평가

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Evaluation of Digital Hearing Aid Algorithms Using Simulation of the Sensori-neural Hearing Losses

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INTRODUCTION

Digital hearing aids offer many advantages over conventional analog hearing aids. Many digital hearing aid algorithms have been developed to compensate hearing losses for hearing impaired listeners. With advances in very large-scale integration, some digital hearing aids seem to reach our minimal requirement as practical hearing aids; cosmetically acceptable size and low power consumption. But a digital hearing aid contains several factors to be considered such as new fitting method and prescription rules tha t should be proven clinically. In the analog hearing aids, there are only few parameters which audiologists can control after determination of hearing aid's hard-wired amplifier. But we have much more choice in the digital hearing aids when fitting into the hearing impaired. In this work, we propose an indirect way to evaluate and predict the performance of the hearing aid systems by using a Hearing Impairment Simulation (HIS). With this objective method, we compared the simulation results with auditory test data of sensori-neural hearing impaired listeners. Then, three currently available digital hearing aid algorithms were evaluated. We also implemented in real-time by using DSP boards and whose performances were evaluated.

Hearing Impairment Simulator (HIS)

The simulation system provides a normal listener with a way to experience the feeling of hearing perception loss that happens to impaired listeners. The HIS consists of a 32-bit floating-point general purpose DSP (Digital Signal Processor) board installed in a host IBM compatible PC. The input signal is sampled at a rate of 16 k Hz, segmented into short time blocks with 50% overlap. The frequency and spectrum of the signal are calculated by using short-time FFT (Fast Fourier Transform). Then, from the measured frequency spectrum , hearing levels in critical bands are calculated. The hearing loss gain of each critical band is calculated by using HLF (Hearing Loss Function). HLF is a function which transforms the dynamic range of normal person to that of the impaired persons. A frequency sampling filter is designed to match the hearing aid characteristics to the impaired listener audiogram. This filtered signal is sent to a reconstruction filter and a DAC.

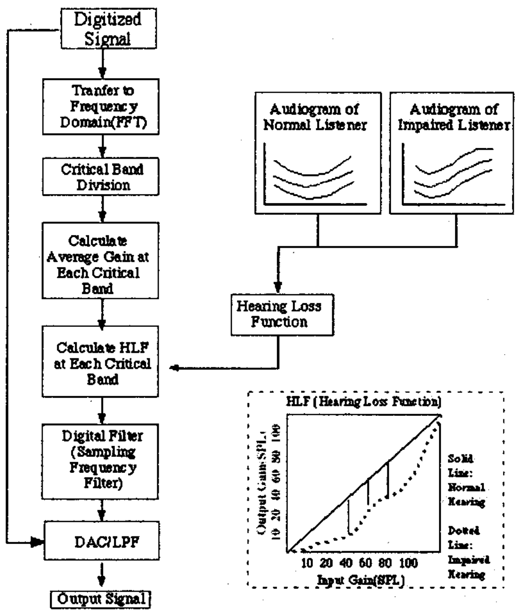

Figure.1 Block diagram of Hearing ImpairmentSimulation

Nonlinear Multiband Compression Algorithms

1) Dynamic Compression Method by a Single FIR Filter

This algorithm determines the gain in each frequency band in such a way that the hearing perception of the impaired listener is restored close to that of normal listeners. We implemented this algorithm both on the Ariel's floating-point DSP board (PC based plug-in type) and Samsung's fixed-point DSP board (body worn type). In this system, we used psychoacoustic factors to compensate the loss of hearing in the manner of summation of loudness and adaptive frequency-dependent amplification.

2) 5-Channel Filter-bank Compression Method Using FIR and IIR Filters

We also implemented 5 channel filter bank hearing aid algorithms using both FIR and IIR filters on the general purpose DSP board. These two systems use the same gain function used in the dynamic compression method described previously.

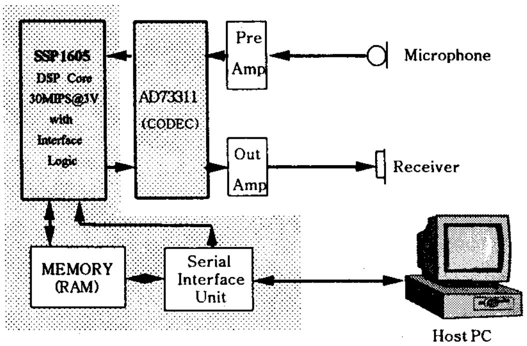

Figure.2 System configuration of the digital hearing aids implemented by using a SSP1605 DSP processor.

EVALUATION EXPERIMENT

For the performance evaluation of the HIS and the hearing aid algorithms, we have utilized 14 patient diagnoses with sensori-neural hearing losses and 16 normal persons for the HIS simulation. The utilized equipment was a audiometer GSI10, DAT and Telefonics TDH49 with ANSI S3.6-1989 calibration.

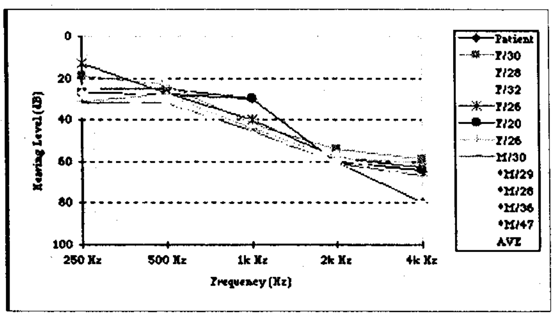

Table.1 Hearing thresholds of normal listeners with HIS.

* Listeners with C5 dip or high frequency hearing losses.

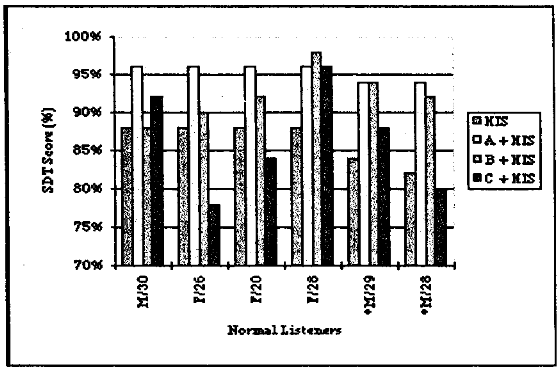

Table.2 SDT(Speech Discrimination Test) scores for each aid algorithms with HIS

HIS : Hearing Impairment Simulator,
A + HIS: Dynamic compression method using a single FIR Filter with HIS,
B + HIS: 5 Ch. filter bank FIR filter with HIS,
C + HIS: 5 Ch. filter bank IIR filter with HIS

Although all fittings vary depending on the specific configuration of the hearing loss, we used three basic steps to set parameters for the digital hearing aid. First, we chose one of basic prescription rules employed in conventional hearing aids such as POGO or NAL-R, and then adjusted fitting for each subjects with real-time measurement. Last, we used speech test to evaluate the hearing aids, such as SDT and SRT (Speech Reception Threshold).

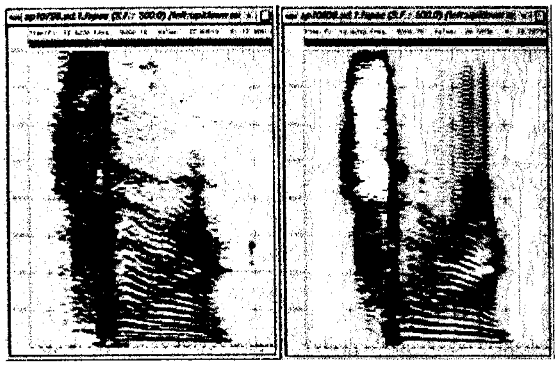

Figure.3 Spectrograms of Korean speech 솔 (s o : l) and enhanced speech using dynamic compression method with a single FIR filter.

CONCLUSIONS

We proposed an indirect way to evaluate and predict the performance of the hearing aid systems, and evaluated the performance of digital hearing aid algorithms using Hearing Impairment Simulation along with subject test for verification. Results have shown that the Dynamic Compression Method is superior to the other algorithms in terms of the ability to understand speech.

The hearing impairment simulator has demonstrated its usefulness for evaluating the performance of hearing aid algorithms, although it is still necessary to continue investigating and evaluating the hearing impairment simulator and the proposed digital hearing aid. To verify the better fitting methodology of the digital hearing aid, more subject tests with real impaired persons as well as normal persons wearing HIS are currently being undertaken.

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