DSP-IMPLEMENTATION OF A MULTIBAND LOUDNESS CORRECTION HEARING AID

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ABSTRACT

A new multiband loudness correction algorithm for digital hearing aids has been implemented on a digital signal processor (TMS320C25, TI). The short-time amplitude spectrum of the input signal is amplified and compressed according to patient specific lookup tables. Loudness scaling at different frequencies with normal hearing and hearing impaired subjects is performed to obtain frequency and intensity specific gain factors. The loudness of speech signals is estimated using a psychoacoustic model approximation at the frequency bands of maximum energy concentration. After transformation of the signal into the residual auditory field of the hearing impaired subject, the processed signal is perceived at a natural loudness. Speech test results with 5 hearing impaired subjects using their own conventional hearing aid and the new MLC-algorithm are presented.

INTRODUCTION

Different authors have reported various implementations of a multichannel signal processing strategy to correct loudness perception of hearing impaired subjects. Their common goal is to transform more of the important portions of the acoustical information into the residual auditory field of the hard-of-hearing.
The individual methods differ from each other by several parameters (number of channels, gain- and compression factors, compression threshold, time or frequency domain algorithm). The algorithm presented here, can be characterized as follows:

- Fitting of parameters (gain and compression) by measuring the individual auditory field of the hearing impaired subject relative to normal hearing subjects (loudness scaling).
- Multichannel strategy in the frequency domain

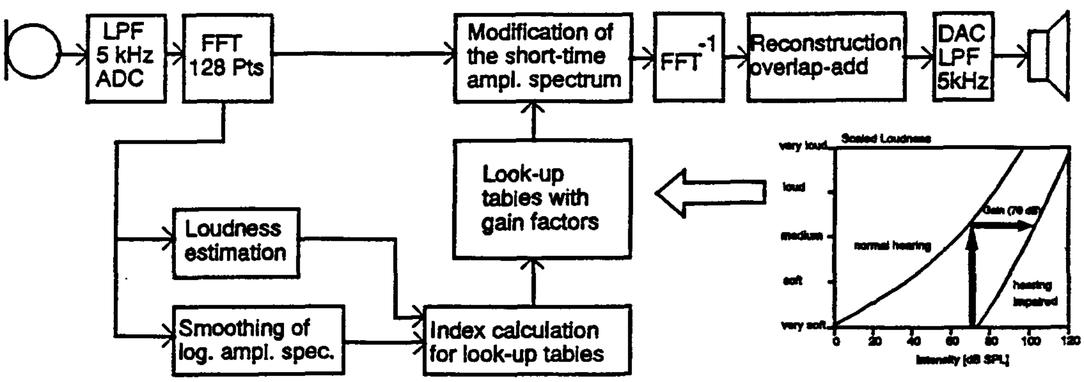

Fig. 1. *Block diagram of the MLC-algorithm with interpolated loudness scaling functions of normal hearing and hearing impaired subjects.*

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(block algorithm).

- Loudness estimation of the incoming signal to determine the required gain which yields the same loudness for the hearing impaired subject as for normal hearing subjects.
- Preventing spectral flattening via control of the compression by a smoothed spectrum.

DESCRIPTION OF THE ALGORITHM

The signal is sampled at 10 kHz and analyzed every 6.4 ms in blocks of 128 samples with a real valued FFT (Fig. 1). In the frequency domain (64 spectral components) the short time amplitude spectrum is modified according to the subject's hearing loss. Reconstruction of the time signal is done by a FFT^{-1} and ovelap-add procedure [1]. The patient-specific lookup tables are determined by performing loudness scaling at 8 frequencies in steps of half-octaves with the hearing impaired and normal hearing subjects. Results are interpolated for the 64 channels.

A simple model estimates the loudness of the incoming signal. It is based on the results of psychoacoustical experiments which investigated the relation between perceived loudness of a single sinusoid sound and a sound with a complex spectrum (e.g. broadband noise) [4]. The model has as output a frequency and an intensity. A sinewave at this frequency and this intensity would be perceived as loud as the complex sound. The frequency is determined as the spectral gravity point and the intensity as the sum of all band intensities. Normally the spectral gravity point lies near the highest energy concentration in the spectrum. Thus, gain tables indices of channels with a high energy concentration are found. Indices for channels with low energy are determined via the smoothed logarithmic amplitude spectrum after correction by the spectral gravity index. By this procedure a flattening of the spectrum is prevented but channels with low energy concentration are still brought into the audible range. The modification of the amplitude spectrum is performed by multiplying real and imaginary part with the gain factors preserving the phase spectrum.

RESULTS

Speech tests with 5 hearing impaired subjects were performed. Each test consisted of 50 digitally recorded german bisyllabic words (VCV and CVC) presented in a 4AFC task. The test items were presented in a sound-treated room at a level of 60 dB SPL either via loudspeaker (with the conventional hearing aids) or monaurally via headphones (with the MLC-algorithm).

Fig. 2 shows the results. All patients reached significantly better discrimination scores with the new MLC-algorithm. Consonants were discriminated 10-50 % better than with their own hearing aid. For vowel tests improvements were smaller than for the consonant test but still remarkable (0-20%).

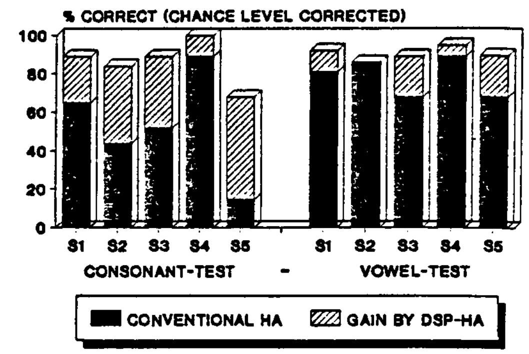

Fig. 2. *Speech test results of 5 hearing impaired subjects.*

CONCLUSIONS

The MLC-algorithm is able to significantly improve speech discrimination for hearing impaired subjects compared to their conventional hearing aid. This is achieved by transforming the input signal into the residual auditory field. Through ongoing loudness estimation weak signals are made audible without over-amplifying strong signal parts. Spectrum flattening is successfully prevented by controlling the compression independently of the fine structure of the spectrum. The implementation of the algorithm on a DSP in connection with a PC allows the use of running speech for quick and flexible fitting of all parameters. In the near future, the algorithm will be implemented in a wearable DSP hearing aid for evaluation of its effectiveness in daily live.

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