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An optimum algorithm for adaptive filtering on acoustic echo cancellation using TMS320C6713 DSP

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ABSTRACT

In this study, it is aimed to enhance the intelligibility of speech by canceling out the echo noise. For this purpose, the data transfer software, which is necessary for real time processing of voice signals, and the adaptive filtering algorithm software for the application of acoustic echo cancellation have been developed. An algorithm has been proposed for the determination of optimum adaptation rate (μ) for the least-mean-square (LMS) adaptation algorithm that is used in the adaptive filter. The effectiveness of our optimum μ value determination algorithm was demonstrated on a single direction voice conference application with one speaker. In this study, we used a DSP card (TMS320C6713), a Laptop computer, an amplifier, a loudspeaker and two microphones in two applications. In the first application, two microphones were placed close to the loudspeaker, while in the other application, one microphone was placed close to loudspeaker and speech trial was implemented in the far-end microphone. Output of the adaptive filter was observed for μ values of 0, 0.1, 100 and optimum (a value between 0.01 and 100). The best results in the adaptive filter were obtained from optimum μ value.

1. Introduction

The audio input to a microphone may contain the speaker's voice and a set pf feedback components from a remote amplified loudspeaker and from surface reflections containing delayed and attenuated versions the speaker's voice. The acoustic feedback components may support sustained decaying echoes which are perceived as annoying echo-chamber like reverberation. These sustained echoes significantly degrade speech quality and speech intelligibility. The most extreme, and annoying, effect of the acoustic feedback path, is a sustained, non-decaying, component known as howling.

Acoustic echo control is a enormously claiming application area. Designing adaptive filters is an important task. The problem of acoustic echo control arises wherever a loudspeaker and a microphone are placed within an enclosure in a way that the microphone pick up the signal from the loudspeaker as well as the reflections from the borders of the enclosure and the various objects inside [1].

The application of acoustic echo cancellation has gained considerable attention in the last decade mainly for the applications of hands-free telephone and tele-conference systems. The main problem in this type of system is the speech signal feedback between loudspeakers and microphones, that is, the acoustic echo which needs to be canceled [2–4]. In the application of acoustic echo cancellation, least mean-square (LMS) algorithm is commonly used on adaptive filtering [1,2,5–9].

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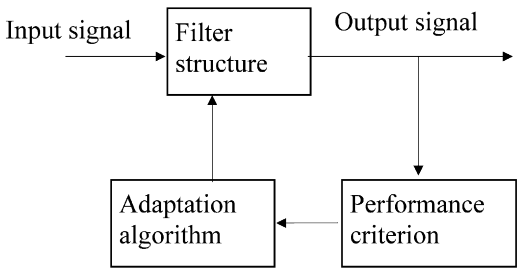

Fig. 1. General structure of adaptive filter.

Echo signal is the delayed form of original speaker signal. That means, echo signal can be assumed as a noise in speaker signal. The removing of noise from the speaker signal cannot be implemented by classical filters, which suppress certain frequency components and pass the others. For this reason, filter design used to remove echo is the subject of optimal filter design. The basic purpose of optimal filter design is to minimize the difference between desired response and actual response of filter. Filter response does not only depend on the statistical information, because physical signals' statistical information has usually a varying nature. Therefore, a filter structure, which is modified its response according to the change of error signal, is required to adapt filter coefficients in a way to minimize error signal [7]. Adaptive filter is the solution to this problem.

Adaptive filter is filter with coefficients which are modified periodically in order to attempt meeting some performance criterion, which is usually in the form of some error or cost function minimization [8]. The adaptive noise canceler consists of an adaptive filter that operates on the reference sensor output to produce an estimate of the noise, which is subtracted from the primary sensor output. The overall output of the canceler is used to control the adjustments applied to the tap weights in the adaptive filter. The adaptive canceler tends to minimize the mean-square value of the overall output, thereby causing the output to be the best estimate of the desired signal in the minimum-mean-square sense [7].

Acoustic echo cancellation system with adaptive filtering realized in this study is composed of two components namely hardware and software [10]. Hardware is consisted of microphone, loudspeaker, amplifier and DSP card. Hardware enables real-time implementation of the application. In developed software, the echo signal present in audio signal is canceled with the implementation of an adaptive filter structure. In the system, optimal filter design studies on echo cancellation with adaptive filtering have been carried out in real-time. Signals have been observed by plotting amplitude-versus-time and power spectrum density-versus-frequency graphics.

2. Method

2.1. Adaptive filter

Adaptive filter is consisted of filter structure and adaptation algorithm. Implemented filter structures are finite impulse response (FIR) or infinite impulse response (IIR). In our study, FIR filter structure which is widely employed in applications due to its stability has been designed. We used the LMS adaptation algorithm, which has been widely employed in literature [5–9].

Adaptive filter is a filter that includes adjustable coefficients by means of an adaptive algorithm to make filter response optimal according to the given performance criterion [7]. In Fig. 1, general structure of adaptive filter is shown.

Filter structure: It is a filter algorithm designed to implement needed operation function, which may be FIR or IIR. This block calculates filter output for the input signal. Also, filter coefficients are updated by adaptation algorithm.

Performance criterion: This block detects unwanted signals i.e. echo signal, in the filter output and generates input signal to cancel echo signal in adaptation algorithm.

Adaptation algorithm: This is the essential component of adaptive filter. This algorithm determines the variation of filter coefficients depending on the performance criterion.

2.2. LMS adaptation algorithm

The least-mean-square (LMS) algorithm is a linear adaptive filtering algorithm that consists of two basic processes [7]:

1. A filtering process, which involves (a) computing the output of a transversal filter produced by a set of tap inputs, and (b) generating an estimation error by comparing this output to a desired response.
2. An adaptive process, which involves the automatic adjustment of the tap weights of the filter in accordance with the estimation error.