Application of Adaptive Algorithms for Acoustic Feedback Cancellation in the Hearing-aid

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ABSTRACT

The hearing- aid is a useful and important device for improving the hearing ability. This study proposes an application of adaptive filtering algorithm for improving acoustic feedback cancellation in the hearing-aid communication system. The hearing-aid receiver becomes perplexed profoundly which brings the acoustic feedback interference problems in the hearing- aid without closed-loop completely since the hearing-aid is much tinier than before recent years. According to the progressed technology, the digital signal processor of the hearing-aid improves the acoustic feedback cancellation and do not affect the function of sound amplifying for eliminating sound waves peak. However, conservative adaptive filters have inferior convergence, lower robustness, and longer computational time, so it could not promote the efficiency of the acoustic feedback cancellation. In order to solve these problems, this study proposes an integrated method includes adaptive filter feedback cancellation, and noise cancellation in the hearing- aid. Besides, there are two capable methods to unravel the problems. First, studying the performance of acoustic feedback cancellation is available by using the Kalman filtering algorithm in the hearing-aid communication system. There are four different filters in evaluating the performance. In addition to the Kalman filtering algorithm, the traditional least-mean-square (LMS) algorithm, normalized least-mean-square (NLMS) algorithm, and recursive least square (RLS) algorithm are also compared in this study. The simulative results indicate that the Kalman filtering algorithm is the most efficient performance in the acoustic feedback cancellation for the hearing-aid communication system. Second, this study applies different directional microphones to solve acoustic feedback more effectively.

Keywords : Acoustic feedback cancellation, Adaptive algorithm, Hearing- aid, Kalman filtering algorithm, Directional microphone

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