ADAPTIVE CHANNEL EQUALIZER USING LMS ALGORITHM

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ABSTRACT

The aim of the project is to demonstrate the operation of adaptive filters using the LMS algorithm , which is the most widely used adaptive filtering algorithm. The adaptive filter adjusts its coefficients to minimize the mean-square error between its output and that of an unknown system. The objective of the project is to change (adapt) the coefficients of an FIR filter, to match as closely as possible the response of an unknown system. First the error signal is computed, which measures the difference between the output of the adaptive filter and the output of the unknown system. On the basis of this measure, the adaptive filter will change its coefficients in an attempt to reduce the error. After repeatedly adjusting each coefficient, the adaptive filter should converge; that is, the difference between the unknown and adaptive systems should get smaller and smaller. The step-size directly affects how quickly the adaptive filter will converge toward the unknown system. If it is very small, then the coefficients change only a small amount at each update, and the filter converges slowly. With a larger step-size, the filter converges more quickly; however, when the step-size is too large, the coefficients may change too quickly and the filter will diverge. MATLAB 7.1 is used for simulation purposes.

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