Adaptive LMS Based Channel Equalization

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Abstract—The main objective of this project is to suppress echoes arising from non-line-of-sight (NLOS) components in wireless communication systems. The increasingly popular use of mobile phones and other wireless communication products has provided the basis and demand for developing appropriate echo suppression techniques.

In wireless communications, the channel has a typically long impulse response. Therefore, a ‘long’ filter model is required to reduce the noise or distortion generated. The Standard Least Mean Square (LMS) channel estimator is used as a basis to this project due to its relatively low computational, complexity, good stability properties, and relatively good robustness against implementation errors. The principles of LMS adaptive Equalizer are investigated, which uses the LMS channel estimator to estimate the NLOS components and subsequently suppress these components. All the results are shown through simulations performed in Matlab environment. The results demonstrated that the design approach investigated in this project is a promising alternative for suppressing distortion in wireless communication systems.

Keywords—Least Mean Square, Non-Line-Of-Site, Equalizer, Filter, Echo, Probability of Error, and Signal to Noise Ratio.

I. INTRODUCTION

Adaptive Filter and Echo:
As the field of study in this project will be based on the ‘Adaptive Filter’, it is worth trying to understand the meaning of the terms ‘adaptive’ and ‘filter’ in a very general sense. The adjective ‘adaptive’ can be understood by describing a system, which attempts to adjust itself so as to respond to some phenomenon that is taking place in its surroundings. This is the meaning of adaptation. A set of procedures is needed for the adaptation process. The ‘system’ that carries out and undergoes the process of ‘adaptation’ is called by the more technical, name ‘filter’. The selection of an appropriate adaptation algorithm and filter structure is always governed by the convergence time, which is the time required to meet the final goal of the adaptation process and the complexity in carrying out the adaptation process. In telecommunication systems, the occurrence of echoes tends to reduce the quality of transmission. An echo is defined as a delayed and perhaps distorted version of a previously transmitted signal. In data transmission, echoes of sufficient magnitude will produce increased error rates and subsequently may lead to the necessity of retransmission, which will affect the efficiency of the transmission system. In speech transmission, the magnitude and spectral distortion as well as the echo delay are the factors contributing to reduction in the quality of reception.

Channel Equalization:
An application of adaptive filters named Inverse Modeling, also known as Deconvolution has found extensive use in various engineering disciplines. The most popular application of Inverse Modeling is in communications where an inverse model, which is also called an Equalizer, is used to reduce the channel distortion.

So, I apply The Inverse Modeling in Channel Equalization. The most significant type of distortion is the pulse spreading effect, which results because the channel impulse response is a response that is non-zero over many symbol periods. This distortion causes inter symbol interference (ISI). The presence of additive noise further deteriorates the performance of data receivers. The role of the Equalizer, as a filter, is to resolve the distortion caused by the channel, through rejection or minimization of ISI, while minimizing the effect of additive noise at its output. The equalizer is usually implemented in the form of a transversal filter. Initial training of the equalizer requires knowledge of the transmitted data symbols, or to be more accurate, a delayed replica of them, since they will be used as signal samples for adaptation of the equalizer tap weights. The idea is to obtain the equalizer's output to be the same as the transmitted data symbols. Hence, an initialization period is involved, for the transmitter to send a sequence of training symbols that are known to the receiver. At the end of the training mode, the tap weights of the equalizer would have converged close to their optimal values [1-21]. Thus, from then onwards, the detected symbols can be treated as the desired signals for further adaptation of the equalizer so that possible variations in the channel can be tracked.

II. MATH

Matlab code is used.

III. UNITS

dB (deci-Bells) is a unit of SNR (Signal-to-Noise Ratio).
B. Abbreviations and Acronyms
1) P.e stands for probability of error.
2) S.N.R stands for signal to noise ratio.
3) L.M.S stands for least mean square filters.
4) F.I.R stands for Finite Impulse Response.
5) IS.I stands for Inter Symbol Interference.
6) d.B stands for deci-Bells.
7) N.L.O.S stands for non-line-of-sight.

V. CONCLUSION
The basic structure, operation, and implementation of the L.M.S Based Channel Equalization are described in several chapters in this project. I consider multi paths in wireless communications, which tend to have ‘long’ impulse responses, consisting of many ‘inactive’ or zero regions interspersed by active regions. In this project, I assume a time-invariant channel.

I first implement the algorithm using Least Mean Square (L.M.S) for the estimation of the channel. So, the error can be occurred due to power loss in signal, which is calculated by removing the estimated value from original value that was the estimation of transmitted signal. Then I estimate the power of the channel that was received like mentioned before.

I increased the step size of the channel so that noise or distortion can be minimized. I pass the channel from F.I.R filter, having weights, called filter co-efficients, then the signal is filtered and noise is removed and finally I receive the high power signal in the output. And when S.N.R increases, the probability of error is minimized. The L.M.S Equalizer was shown to be capable of removing the channel distortion when included in the algorithm of Standard L.M.S. Under higher noise level conditions, the L.M.S Equalizer with the Standard L.M.S and Detection Guided L.M.S algorithms failed, while the Equalizer with Detection Guided L.M.S with Tap Decoupling continued to perform well. However, at excessively high noise levels, this latter Equalizer also failed. It has also been confirmed that the correlation factor does not affect the performance of the L.M.S Equalizer when a long training sequence is allowed. However, when short training sequences are required, a larger correlation factor leads to poorer L.M.S Equalizer performance. Based on the simulation results produced, it is proven that the noise factor affects the asymptotic error, but not the convergence rate. I also found that the correlation factor affects the convergence rate, but not the asymptotic performance.

In conclusion, this project has demonstrated in both theory and Matlab simulations, that the design approach investigated is a promising alternative for suppressing echoes in wireless communication systems.

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