ISSN No.: 2455-6130
Volume 1, Issue 6, 2016, pp. 06-10

website: www.ijoetr.com

International Journal of Emerging Trends in Research

Comparison of Performance of Adaptive Channel Equalizer using LMS, RLS, KALMAN Algorithm

Himanshi Suri[1], Parveen Khanchi[2]

[1, 2]Department of ECE, RIMT, Chidana, India,

Abstract

In non-ideal communication channel, signal quality fades with distance. Moreover, effect of multipath propagation causes transmitted signal to be attenuated, superimposed and delayed at receiver in practical communication channel, causes Inter Symbol Interference (ISI). This ISI is overcome using Channel Equalizer and technique used is Equalization. In digital communication, Adaptive Equalizer which automatically adapts to time-varying properties (parameters) of communication channel. Equalizer is adaptive i.e. its weights are adjusted itself according to the input signals. Adaptive Algorithms such as LMS, RLS and KALMAN are used in digital processing applications. This paper aim to study the performance of LMS, RLS and KALMAN Algorithm for Channel Equalization by making different models using Input signal random integer and output is compared through SIMULINK tool of MATLAB.

Keywords: LMS (Least Mean Square), RLS (Recursive Least Square), MATLAB

1. Introduction

The requirement is to design a reliable system so that data is transmitted at higher rate. ISI and thermal noise affect the data in channel. When designing optical receiver in various applications, receiver filter requires functions which can estimate the interference. Interference statistics should be known a priori for filter design. Adaptive filter adaptively estimate the interference. Adaptive algorithm can be applied in any application where an adaptive filter may be needed. In LMS algorithm, both converges speed and residual error level are decided using step size. RLS algorithm has fast converges and high complex operations per sample. KALMAN filter has good performance in quality of estimation but high complexity in operation.

2. Algorithms of adaptive channel equalizer

The different adaptive algorithms used for designing channel equalizer in this thesis are

- LMS algorithm

- RLS algorithm
- Kalman algorithm

LMS algorithm uses the estimates of the gradient vector from the available data. LMS incorporates an iterative procedure that makes successive corrections to the weight vector in the direction of the negative of the gradient vector which eventually leads to the minimum mean square error. The LMS algorithm results can be summarized as

- Output, y(n)= w(n)x(n)
- Error, e(n) = d(n) – y(n)
- Weight, w(n+1) = w(n) + μ x(n)e(n)

Where x (n) is the input signal applied and μ is the step-size parameter and controls the convergence characteristics of the LMS algorithm.
RLS algorithm is adaptation algorithms are based on the exact minimization of least-squares criteria. The filter weights are optimal at each time instant n. Adaptive RLS algorithms are the time-recursive analogs of the block processing methods of linear prediction and FIR Wiener filtering. Steps of RLS algorithm are
compute the error signal $e_{n=}\ x_n$-$\hat{x}_n$
adjust the weights h(n)=h(n-1)+e(n/n-1)k(n)

- k(n) is calculated as k(n)=u(n)k(n/n-1)
- u(n)=1/[1+v(n)]
- v(n)= $k(n/n-1)^T$ y(n) , k(n/n-1)= f^{-1} P(n-1) y(n)
- P (n) = $R(n)^{-1}$
- h(n) = $R(n)^{-1}$r(n) , R(n)= $\sum_{k=0}^{n} f^{(n-k)} y(k) y^T(k)$
- r (n) =$\sum_{k=0}^{n} f^{(n-k)} x(k) y(k)$

KALMAN Algorithm is an algorithm that uses a series of measurements observed over time. The algorithm works in a two-step process. In the prediction step, the Kalman filter produces estimates of the current state variables, along with their uncertainties. Once the outcome of the next measurement is observed, these estimates are updated using a weighted average.
The Kalman adaptive algorithm can be summarized as:

- Initialize by $\hat{x}_{0/-1}$ = 0 and $P_{0/-1}$=E [x_0^2].
- At time n,$\hat{x}_{n/(n-1)}$ and $P_{n/(n-1)}$ and the new measurement Y_n are available.
- Compute $\hat{y}_{n/(n-1)}$=c$\hat{x}_{n/(n-1)}$ and the gain G_n using relation $G_n=\frac{E[\epsilon_n x_n]}{E[\epsilon_n^2]} = \frac{cP_{n/(n-1)}}{R+c^2P_{n/(n-1)}}$
- Correct the predicted estimate $\hat{x}_{n/n}$=$\hat{x}_{n/(n-1)} + G_n \epsilon_n$ and compute its mean-square
- $P_{n/n}$, using relation
- $P_{n/n}$=$P_{n/(n-1)}$-G_nc$P_{n/(n-1)}$ $= P_{n/(n-1)}$-$\frac{c^2P_{n/(n-1)}}{R+c^2P_{n/(n-1)}}$=$\frac{RP_{n/(n-1)}}{R+c^2P_{n/(n-1)}}$
- Predict the next estimate $\hat{x}_{(n+1)/n} = a\hat{x}_{n/n}$ and compute the mean-square prediction error $P_{n+1/n}$ using relation $P_{n+1/n} = a^2P_{n/n} + Q$
- Go to the next time instant, n → n + 1.

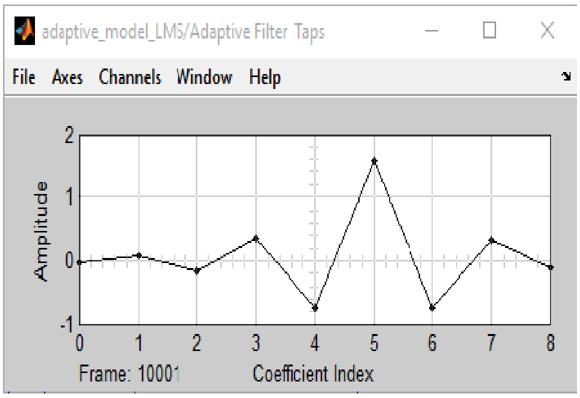
The optimal predictor $\hat{x}_{n/(n-1)}$ satisfies the Kalman filtering equation, i.e.

$\hat{x}_{(n+1)/n}$=a$\hat{x}_{n/n}$=a ($\hat{x}_{n/(n-1)}$+$G_n\epsilon_n$)=a$\hat{x}_{n/(n-1)}$+a G_n(y_n-c$\hat{x}_{n/(n-1)}$).

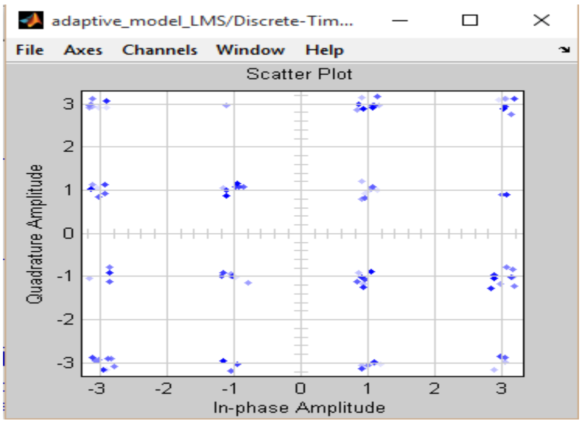
3. Results

Simulation results of different models of channel equalizer based on different algorithms providing random input signal as its weight are adjusted itself.

- *Simulation result of LMS Algorithm*

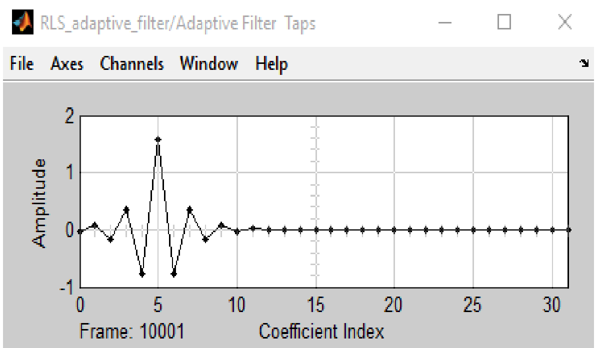


Graph 3.1 Amplitude vs Coefficient Index

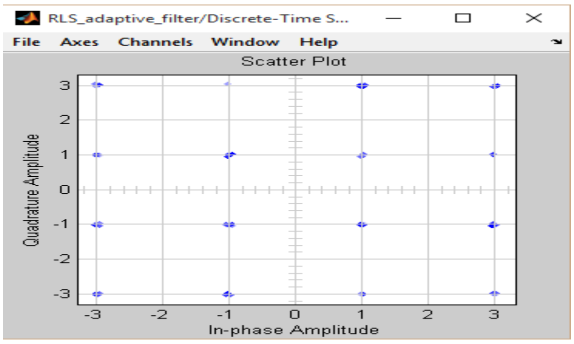


Graph 3.2 Quadrature vs Inphase Amplitude

- *Simulation result of RLS Algorithm*

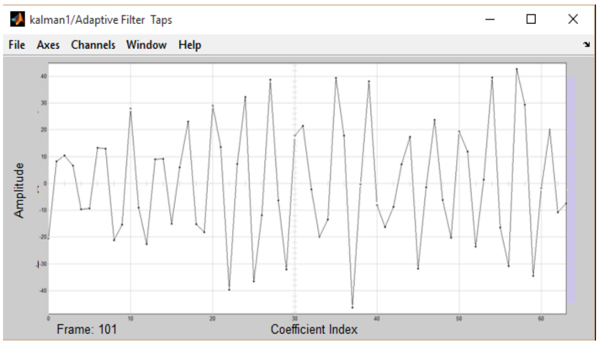


Graph 3.3 Amplitude vs Coefficient Index

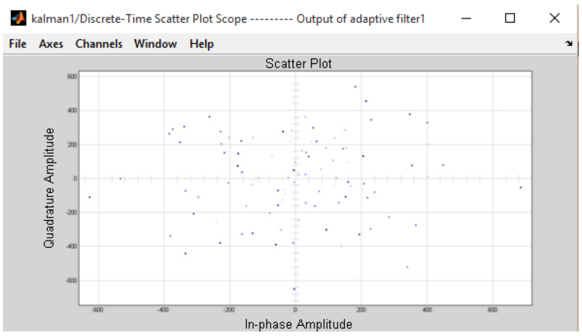


Graph 3.4 Quadrature vs Inphase Amplitude

- *Simulation result of KALMAN Algorithm*



Graph 3.5 Amplitude vs Coefficient Index



Graph 3.6 Quadrature vs Inphase Amplitude

4. Conclusions

1. RLS algorithm is the best among the three algorithms. On the basis of the error graphs on the adaptive RLS filter output side; we can conclude that the error possibilities correction and error occurrence possibilities are very less in case of RLS filtering.

2. LMS algorithm is good. The output of adaptive LMS filter takes a bit of time to get settled with a bit slow rate. The magnitude of the error are a bit high in case of LMS as compared to RLS filtering.

3. Kalman algorithm is the worst. Performance of the Kalman filter is worse among all three adaptive filters, the error rate is very high on output side.

Acknowledgment

It’s a great opertunity to express the gratitude to have a mentor and guide like Er. Mohinder Malhotra. He provide me all the necessary guidance and assists me finding solutions of various problems.

References

Krstajic, B., Uskokovic, Z., & Stankovic, L. (2003). Adaptive channel equalizer with new VSS LMS algorithm. In ISSPA (2) (pp. 567-570).

Cheng, Y. H., Lu, Y. H., & Liu, C. L. (2006, June). Adaptive channel equalizer for wireless access in vehicular environments. In ITS Telecommunications Proceedings, 2006 6th International Conference on (pp. 1102-1105). IEEE.

Leopold, R. J. (1992, November). The Iridium communications systems. In Singapore ICCS/ISITA'92.'Communications on the Move' (pp. 451-455). IEEE.

Lee, I., & Jenkins, W. K. (1997, August). VLSI design for an adaptive equalizer using a residue number system architecture for magnetic channels. In Circuits and Systems, 1997. Proceedings of the 40th Midwest Symposium on (Vol. 2, pp. 782-785). IEEE.

Haykin, S. S. (2008). Adaptive filter theory. Pearson Education India.