Major Project Presentation
on
CHANNEL EQUALIZATION USING ADAPTIVE ALGORITHM

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Project Guide:

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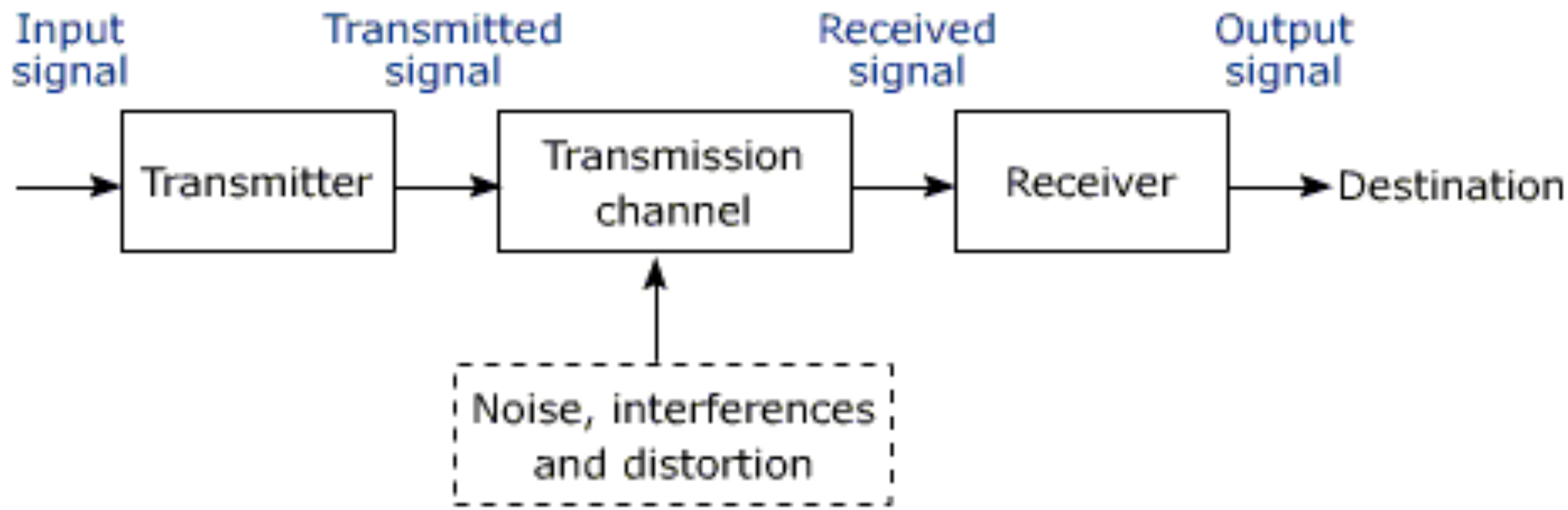
OUTLINE

- OVERVIEW OF COMMUNICATION SYSTEMS
- NEED OF CHANNEL EQUALIZATION
- PROBLEM FORMULATION
- ADAPTIVE FILTERING APPROACHES
- CHANNEL EQUALIZATION AND ADAPTIVE ALGORITHMS
- SIMULATION RESULTS
- CONCLUSION
- SCOPE FOR FUTURE WORK

OVERVIEW OF COMMUNICATION SYSTEMS



MOTIVATION

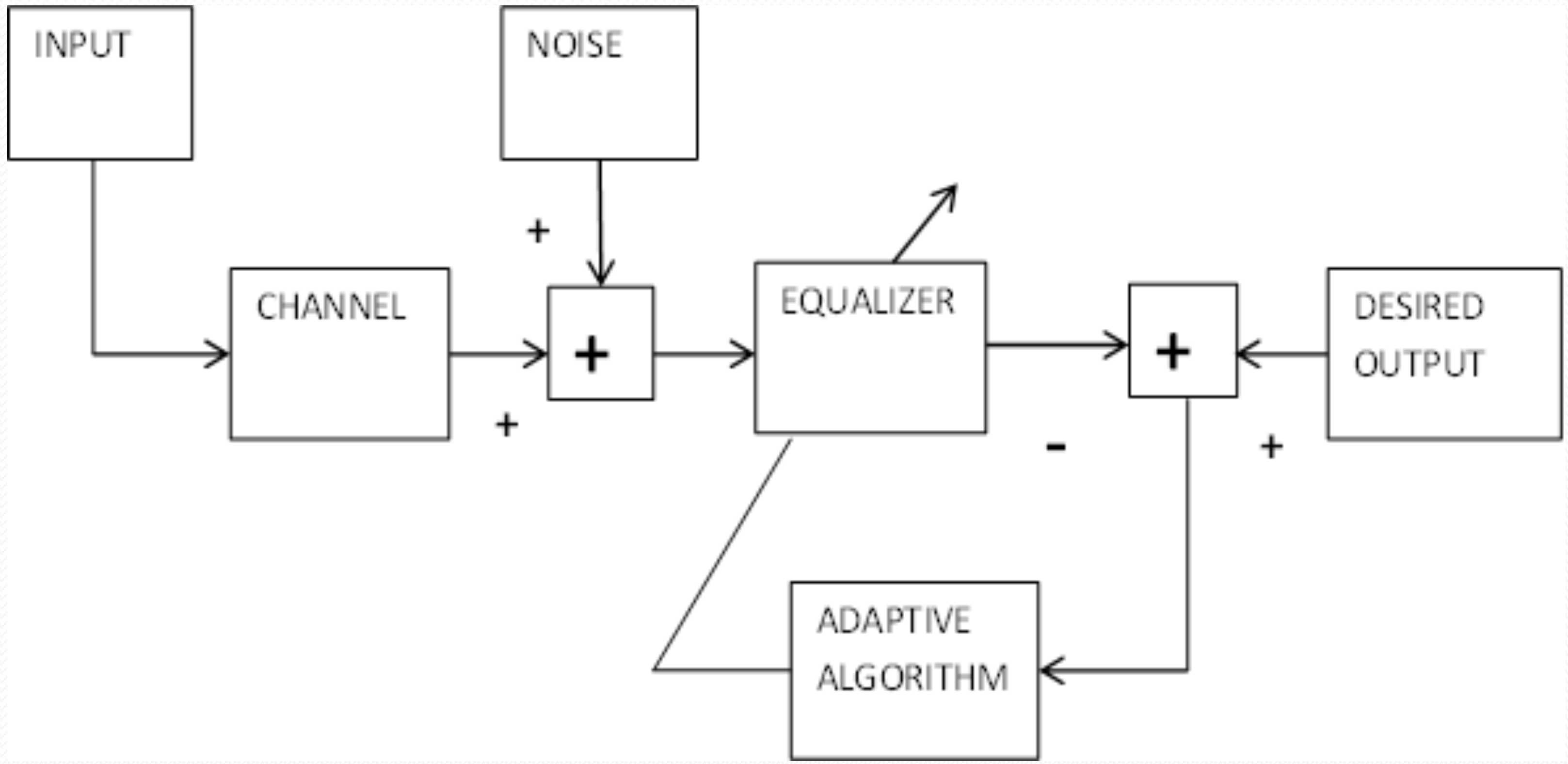
- To compensate channel impairments like ISI(Inter symbol Interference) and additive noise which are prominent in the dispersive wireless channel.
- This will require the design of adaptive channel equalizer to estimate the original pulse transmitted in binary format.

PROBLEM FORMULATION AND CHANNEL MODELING

- The problem dealt here is the equalization of an unknown channel.
- The channel is generally modeled as a FIR filter with constant and variable coefficients.
- PAM Constellation is considered in presence of White Gaussian Noise.
- The FIR Filtering model is given as

$$y(n) = \sum_{i=o}^{N} b_i x(n-i)$$

CHANNEL EQUALIZATION MODEL



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LINEAR EQUALIZER:-

- Aims at minimizing the difference between the transmitted data and the Equalizer output.

DECISION FEEDBACK EQUALIZER:-

- It is a non linear equalizer that uses the previous detector decision to eliminate the ISI(Inter Symbol Interference) on pulses that are currently being modulated.

ADAPTIVE FILTERING APPROACHES

- The entire project work is based on equalization of fixed and variable coefficients of the unknown FIR channel.
- The noise corrupted signal is then passed through the linear and decision feedback equalizer which is trained by using adaptive algorithms like LMS(Least Mean Square), NLMS(Normalized Least Mean Square), RLS(Recursive Least Square) and UKF(Unscented Kalman Filter).
- The BER(Bit error rate) Vs SNR(Signal to noise ratio) along with the MSE(Mean Square Error) is plotted.

LMS AND NLMS APPROACHES

- The update equation for the LMS algorithm is

$$w(n+1) = w(n) + \frac{1}{2}\mu[-\nabla(E\{e^2(n)\})]$$

which is derived from standard deviation as an approximation and the step size μ is originally considered for a deterministic gradient.

- The NLMS algorithm is given by:

$$W(n+1) = W(n) + \frac{x(n) * e(n)}{x^T(n)x(n)}$$

RLS AND UKF APPROACHES

- The weight updated equation in RLS is given by

$$W(n+1) = W(n) + k(n) * e(n) \qquad k(n) = \frac{p(n) * u(n)}{\lambda + u(n) * p(n) * u^T(n)}$$

where k(n) is the gain vector, p(n) is input correlation vector and λ is the forgetting factor.

- The Kalman Estimation is based on

$$x_K = f(x_{k-1}, k-1) + q_{k-1} \quad \text{and} \quad y_k = h(x_k, k) + r_k$$

where q_k is the process noise and r_k is measurement noise. The weight updating equation is given as

$$W_K = W_k^- + K_k[y_k - \mu_k] \qquad P_k = P_k^- - K_k S_k K_k^T$$

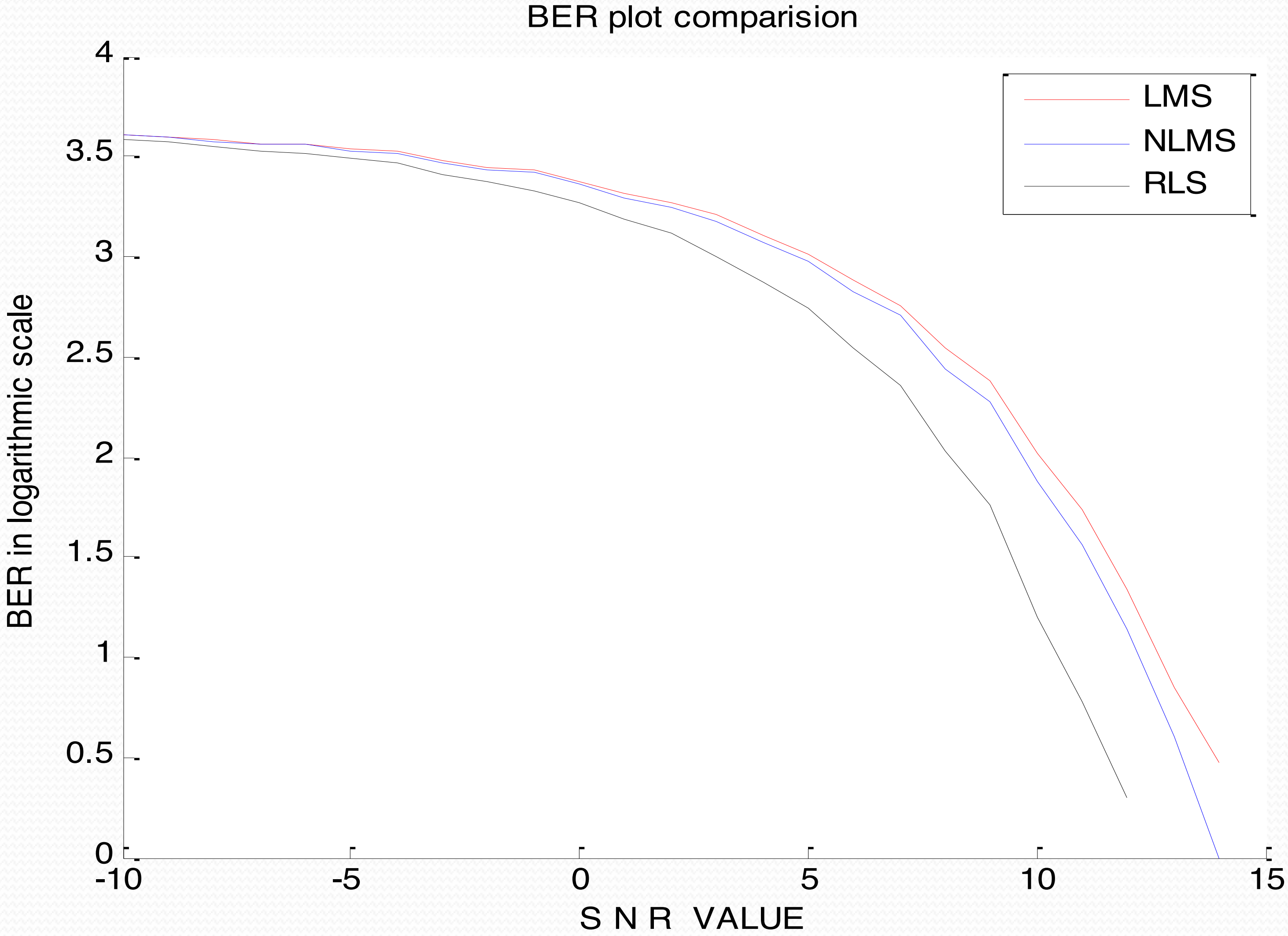
Simulation Results and Discussions

- All the simulations have been performed using MATLAB 7.7.0 (R2008b).The input used is a 2000 sample PAM(Pulse Amplitude Modulation) sequence which is corrupted by White Gaussian Noise of 20db. The SNR(Signal to noise ratio) is varied from -10db to 15db to plot the BER(Bit Error Rate) plot.
- Case 1:- Constant Coefficient channel used:

$$h(n)= 0.2\delta(n) + 0.3\delta(n-1) + \delta(n-2) + 0.3\delta(n-3)$$

Comparison of BER Vs SNR Plot of 4 tap FIR channel using different Adaptive Algorithm is shown in subsequent slide.

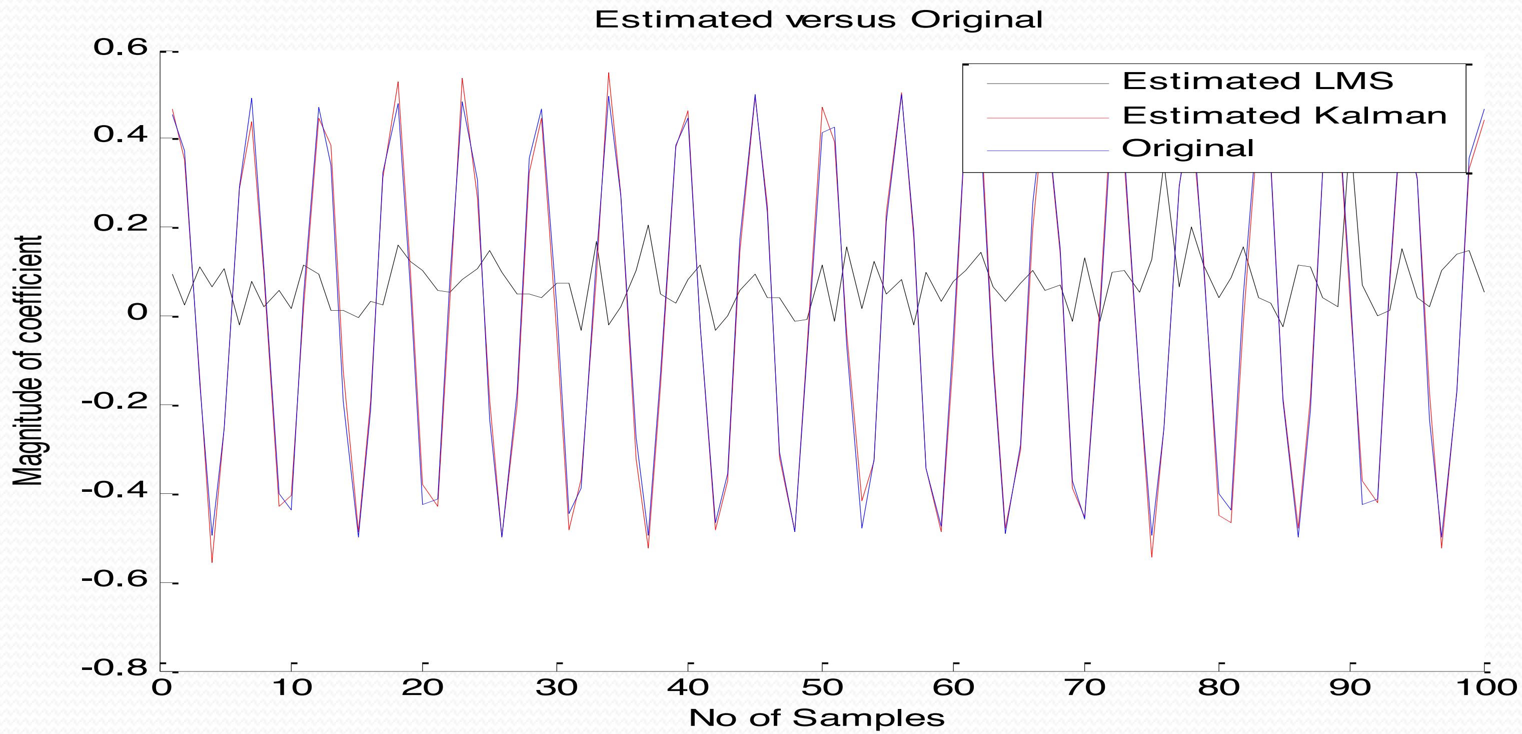
BER PERFORMANCE CURVE LMS, NLMS, RLS



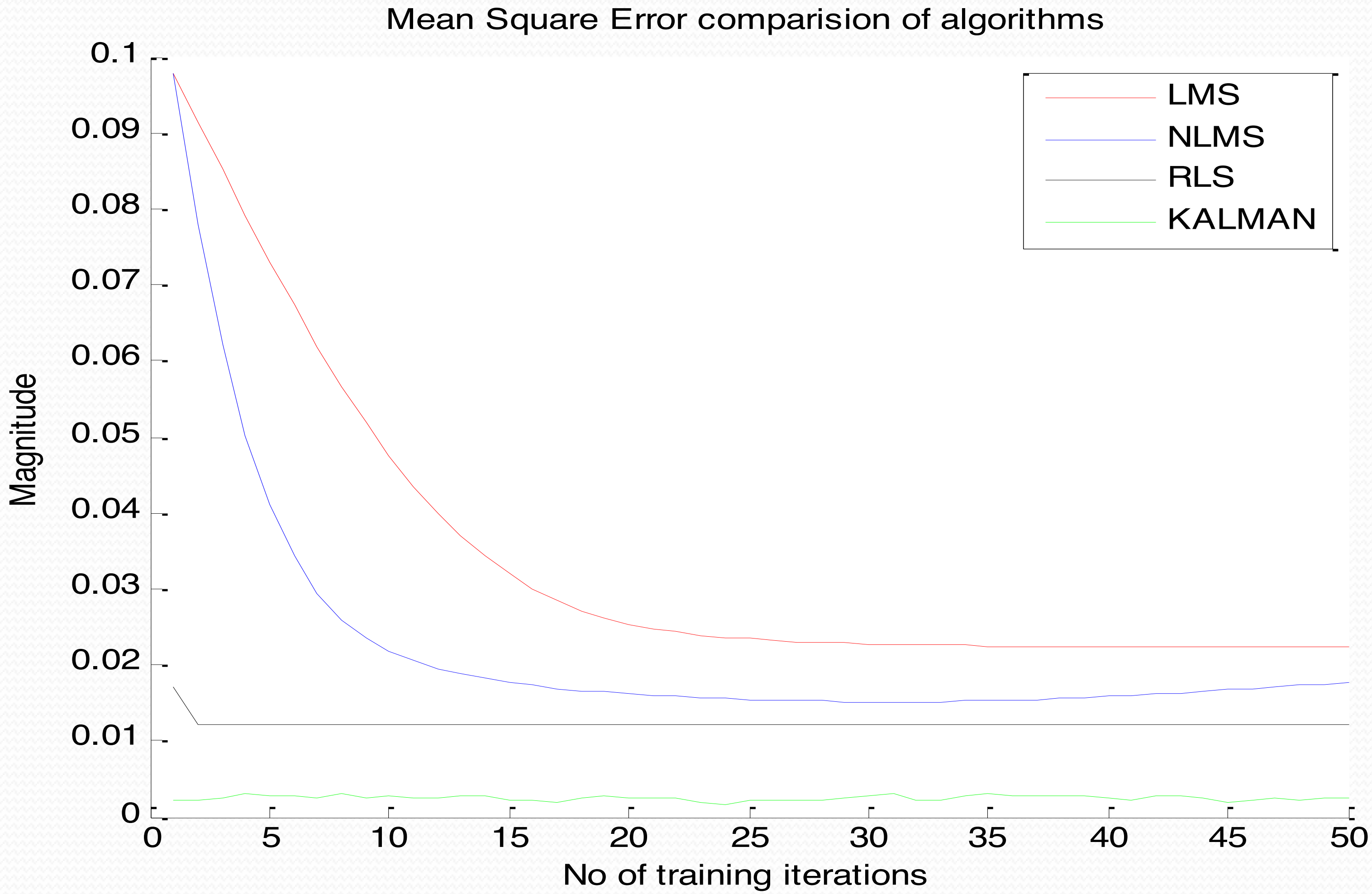
- Case 2 :- 4 tap FIR filter using Time varying Coefficient channel is modeled as

$$h(n)=[0.6\cos(50n)\ ,\ 0.8\sin(60n),\ 0.5\sin(20n),\ 0.1\cos(10n)]$$

Coefficient tracking of Time varying channel LMS versus KALMAN is shown below



Case 3:- MSE(Mean Square Error) plot of LMS NLMS,RLS and UKF



CONCLUSION

- LMS and NLMS:

The LMS algorithm has low convergence ratio and has poor tracking capability in case of time varying and nonlinear channels.

- RLS:

It has high or more computational complexity due to calculation of lemma as compared to LMS but has relatively good convergence properties. If the actual channel model doesn't match with the analytical model used for estimation , the filter will fail to track the time variation properly.

- UNSCENTED KALMAN FILTER:

It has superior performance over the other algorithms as it is a derivative free algorithm. It has less computational complexity and can track the time varying coefficients of the channel accurately when the system dynamics is highly nonlinear.

SCOPE OF FUTURE WORK

- Input: QPSK, QAM and BPSK signal constellation may be used.
- Filter: Particle filter , Square root Quadrature Kalman Filter for better tracking.
- Structure: Artificial Neural network based equalization model
- Channel: Standard models like Rayleigh’s and Rician Fading Channel
- MIMO system can be considered for equalization

REFERENCES

- [1] Widrow, B. and Walach, E., *Adaptive Inverse Control,* Prentice-Hall, Upper Saddle River, NJ, 1996.
- [2] Lucky, R.W., Techniques for adaptive equalization of digital communication systems, *Bell Sys. Tech. J.,* 45, 255–286, Feb. 1966.
- [3] S. Haykin, Adaptive Filter Theory, third ed., Prentice-Hall, Upper Saddle River, NJ, 1996.
- [4] Haykin. S. Digital Communication. Singapore*: John WilSons Inc, 1988.*
- [5] Qureshi, S.U.H., Adaptive equalization, *Proc. IEEE,* 73(9), 1349–1387, Sept. 1985.c 1999 by CRC Press LLC.
- [6] Kalman filtering theory and applications using matlab: By Grewal and Andrews .
- [7] Kalman filtering implementation, University of Stuttgart, Institute of Geodesy.
- [8] Jian Zhang, Akshya Swain, Nirmal-Kumar C Nair , J. J. Liu, Estimation of Power Quality using an Unscented Kalman Filter.
- [9] P.K. Dash, S. Hasan, B.K. Panigrahi, Adaptive complex unscented Kalman filter for frequency estimation of time-varying signals.
- [10] Weihua Zhuang, RLS Algorithm with Variable Forgetting Factor for Decision Feedback Equalizer over TimeVariant Fading Channels.

THANK YOU ☺

ANY QUESTIONS??