

# **EQUALIZATION OF CHANNEL USING KALMAN FILTER**

A Dissertation Submitted towards the Partial Fulfillment of Award of  
Degree of

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## CERTIFICATE

This is to certify that the work contained in this thesis entitled “*EQUALIZATION OF CHANNEL USING KALMAN FILTER*” by **Vipin Sharma** in the requirement for the partial fulfillment for the award of the degree of **Master of Technology in Microwave and Optical Communication Engineering, Delhi Technological University**, New Delhi is an account of her work carried out under my guidance in the academic year **2011-2012**.

This work embodied in this dissertation has not been submitted for the award of any other degree to the best of my knowledge.

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## **ABSTRACT**

The needs for increasing speeds and capacities in modern data communication systems have led to increasing hopes for the introduction of optical communication in a wide range of communication networks. In communications, a critical manifestation of distortion is inter-symbol interference (ISI), whereby symbols transmitted before and after a given symbol corrupt the detection of that symbol. All physical channels (at high data rates) tend to exhibit ISI. It will be gainful to delve into techniques that can mitigate the effects of ISI and thus improve transmission speed and save bandwidth.

The equalizer attempts to extract the transmitted symbol sequence by counteracting the effects of ISI, thereby improving the probability of correct symbol detection. Its purpose is to reverse the effects that the channel has on the transmitted signal, with the aim of reproducing the original signal at the receiver end. Data based channel estimation methods offer low complexity and good performance and are thus quite widely used in communications systems today. But they are also wasteful of bandwidth since they use training sequences to estimate the channel.

This thesis presents a method of improving the channel estimate without increasing the length of the training sequence. This method uses the underlying channel model and the available data based estimate, to implement the channel estimation algorithm in the form of a Kalman filter. The Kalman filter based channel estimator leads to a significant gain in performance as compared to the data-only estimator. The Kalman filter also allows us to predict the state of the channel before the frame is actually received. In this thesis, the channel is estimated by using a Kalman filter. The channel is time varying modeled as a low-pass tapped delay line filter that works as the FIR filter with time varying coefficients. Here the Kalman filter technique is used to estimate the time varying coefficient of the channel.

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