Adaptive receiver structures for time-dispersive channels
by Baldev Krishan

A thesis submitted to the Graduate Faculty in partial fulfillment of the requirements for the degree of DOCTOR OF PHILOSOPHY in Electrical Engineering
Montana State University

Abstract:
Intersymbol interference is one of the main problems encountered in high-speed data transmission over telephone, radio and undersea channels. This interference, caused by the time dispersion and multipath characteristics of the channels, has been a major hindrance to increased data rates at acceptable error probabilities. Various linear and nonlinear receivers have been proposed to combat the intersymbol interference problem. It has been found by Proakis and Magee that a receiver consisting of a Viterbi decoder in combination with a channel estimator provides a receiver structure which is realizable and asymptotically optimum. The object of this research has been the development of faster adjustment algorithms for the channel estimator so that the estimator can be used on more rapidly varying channels. Forney has proposed a suboptimum algorithm for sequence estimation in the presence of ihtersymbol interference. This suboptimum algorithm is suitable for a particular type of partial response system. Another aspect of this research is to study the sensitivity of Forney's suboptimum algorithm and to develop an adaptive equalization scheme for making Forney's suboptimum algorithm suitable for some other type of partial response system. Simulation results are obtained for this adaptive equalization scheme.

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March, 1974

ACKNOWLEDGMENTS

The author wishes to express his sincere gratitude to his advisor, Dr. C. K. Rushforth, for his active participation and valuable guidance during the course of this research.

The author would also like to thank Dr. D. A. Pierre and Dr. D. N. March for their comments and helpful suggestions.

The author is indebted to Dr. P. E. Uhlrich for the financial support given during the author's stay at Montana State University and to Professor R. C. Seibel for his assistance. The author expresses his appreciation to Mrs. Jean Julian for her excellent technical typing.

Finally, the author would like to thank his parents without whose guidance and support this thesis would not have been possible, and to the author's wife, Tripta, for her patience and encouragement throughout the development of this thesis.

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ABSTRACT

Intersymbol interference is one of the main problems encountered in high-speed data transmission over telephone, radio and undersea channels. This interference, caused by the time dispersion and multipath characteristics of the channels, has been a major hindrance to increased data rates at acceptable error probabilities. Various linear and nonlinear receivers have been proposed to combat the intersymbol interference problem. It has been found by Proakis and Magee that a receiver consisting of a Viterbi decoder in combination with a channel estimator provides a receiver structure which is realizable and asymptotically optimum. The object of this research has been the development of faster adjustment algorithms for the channel estimator so that the estimator can be used on more rapidly varying channels. Forney has proposed a suboptimum algorithm for sequence estimation in the presence of intersymbol interference. This suboptimum algorithm is suitable for a particular type of partial response system. Another aspect of this research is to study the sensitivity of Forney's suboptimum algorithm and to develop an adaptive equalization scheme for making Forney's suboptimum algorithm suitable for some other type of partial response system. Simulation results are obtained for this adaptive equalization scheme.

Chapter I

INTRODUCTION

1.1 Historical Review

It seems inherent in nature that signals are dispersed--that is, spread out and changed--in transmission. This dispersion manifests itself in two ways: in radio communication it is called multipath, and in wire communication it is called linear distortion. Multipath transmission can be viewed as communication through a group of parallel, distortion-free channels of different lengths. The wire communication channel may have a single obvious transmission path but a non-ideal frequency characteristic (an ideal transmission frequency characteristic is here defined as one having a flat amplitude frequency response and a linear phase frequency response). Figure 1.1 displays frequency response curves for such a non-ideal channel. This linear distortion present in wire communication causes what is called intersymbol interference.

Intersymbol Interference:

Transmission of data on voice channels, over the switched telephone network, has increased markedly over the past few years. The data is used to amplitude modulate a train of identically shaped pulses. These pulses are

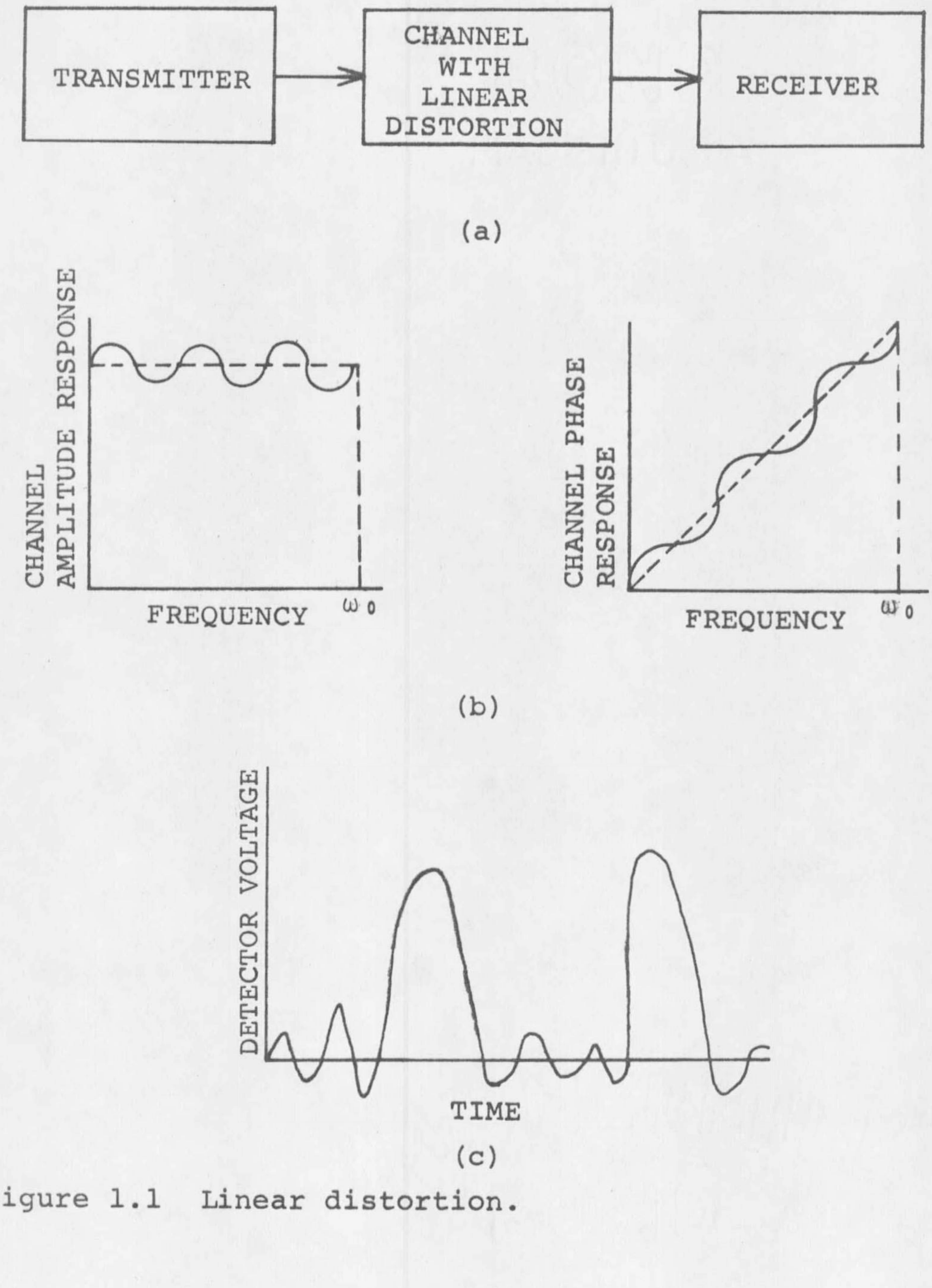

Figure 1.1 Linear distortion.

transmitted over the channel, demodulated by the receiver, and processed. Although the signal-to-noise ratio (SNR) is sufficient to permit higher rates, the non-uniform transmission characteristics of the channel causes what might be termed a distortion barrier prohibiting faster transmission. The distortion of data pulses by the channel results in these pulses being spread out in time so as to overlap other transmitted pulses. The phenomenon of pulse overlap and the resultant difficulty with receiver decisions is termed intersymbol interference. This intersymbol interference is one of the chief degrading factors in present data-transmission system and becomes the determining factor in the design of higher rate system.

Intersymbol interference normally arises in pulse modulation systems. It is the primary impediment to reliable high-rate digital transmission over high signal-to-noise ratio narrow bandwidth channels such as voice-grade telephone circuits. Sometimes a controlled amount of intersymbol interference is introduced deliberately to achieve certain beneficial effects in data transmission. These systems, which utilize a controlled amount of intersymbol interference, are called partial response systems [41-42].

The channel can be represented, as shown in Figure 1.2, by a time-varying impulse response c(t), plus additive white noise. The output of this channel typically differs from its input in two ways. First, the channel may modify the message in a deterministic, although not necessarily known, fashion. Examples of this type of modification are dispersion, frequency offset, and nonlinear action. The channel can also corrupt the transmitted signal statistically. In this category are various types of additive and multiplicative noise such as thermal and impulse noise. Normally, the frequency offset is very small compared to the system bandwidth and is, therefore, neglected. Impulse noise is characterized by long, quiet intervals followed by bursts of much higher amplitudes than would be predicted by a normal or Gaussian distribution law, and, therefore, this noise is usually not considered in the mathematical model of the channel. The model, therefore, accounts for the most deterministic impairment, time dispersion, as well as the most prevalent and mathematically tractable Gaussian noise.

Intersymbol interference represents a kind of deterministic impairment. If the channel characteristics

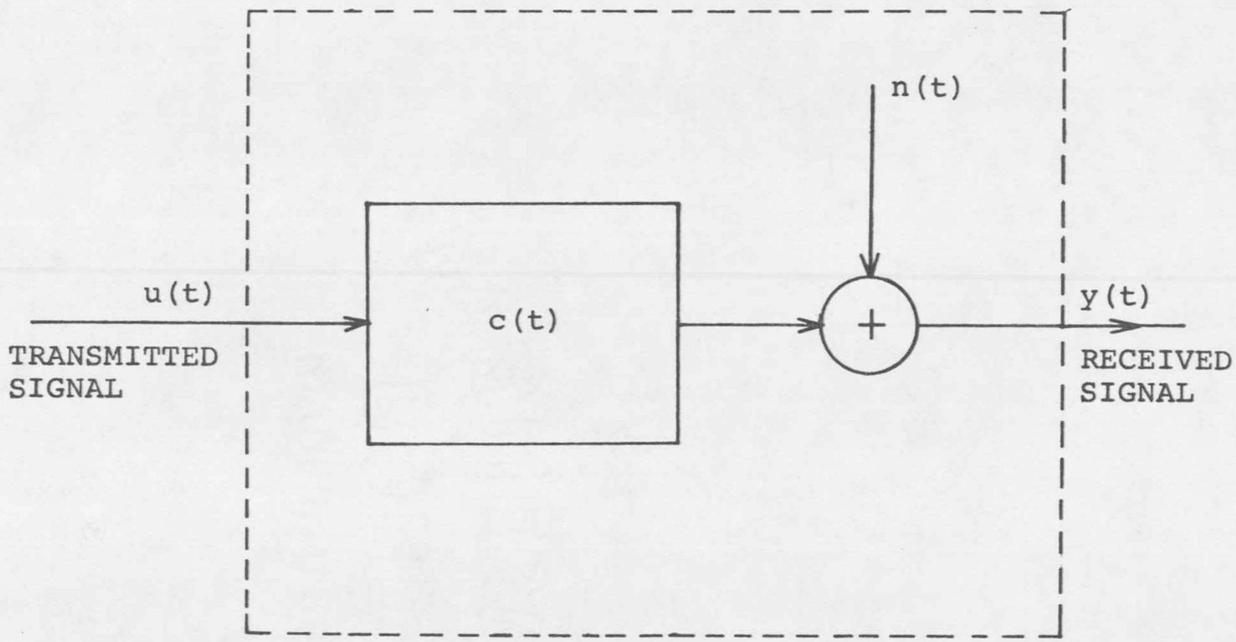

Figure 1.2 Linear time-varying channel model.

are known, it may be possible in theory to remove intersymbol interference. However, the transmission channel is never known exactly; thus, it is impossible to eliminate completely the effects of intersymbol interference in real data transmission systems. There are several methods of combating intersymbol interference. The one which has been quite extensively used in symbol-by-symbol decision systems is that in which the transmitted signals are designed so that all signals except the desired symbol have zero value when the waveform is sampled at the receiver. This method is based on the work by Nyquist [1] and the resulting specification is known as the Nyquist criterion.

In the case of PAM (pulse-amplitude modulation) systems, different criteria have been used to design the optimum receiver in the presence of intersymbol interference. The different criteria used are: (1) Minimization of probability of error caused by intersymbol interference and noise, (2) Minimization of mean-square error resulting from intersymbol interference and noise.

Under either of these criteria, the optimum linear receiver is invariably a matched filter followed by a transversal filter. The matched filter might be interpreted as treating the noise while the transversal filter corrects intersymbol interference. When the signal-to-noise ratio is low, the principal correction must come from the matched filter; for high signal-to-noise ratios, intersymbol interference must be almost completely suppressed and the principal correction effected by the transversal filter.

The approach used in deriving the optimum receiver has been to minimize the effect of intersymbol interference at the receiver. The decision about the input sequence is again made on a symbol-by-symbol basis. Most of the work in the derivation of optimum linear receiver has been done by Tufts, George, Aaron, and Ericson [2-6]. Ericson [5] has also shown that any reasonable criterion of goodness will lead to an optimum filter that has the form discussed earlier. Berger and Tufts [2,4] considered the joint optimization of transmitter and receiver in pulse amplitude modulation using the criterion of total mean-square error. The optimum linear receiver based on receiver filter optimization is shown in Figure 1.3. The transversal filter is shown in Figure 1.4.