

# Reducing Symbol Error Rate Using Adaptive Equalizer

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**ABSTRACT** - MSER equalizer is derived based on the Bit Error Rate or Symbol Error Rate objective function, Due to the complicated structure of the BER/SER objective functions, the derivation of the adaptive MBER/MSER (AMBER/AMSER) equalizers is not straightforward. Moreover, the resulting adaptive equalizers in existing work usually have complicated structures. Present a new approach to derive the AMBER/AMSER equalizers. Consider a sample-by-sample adaptive equalizer, where the equalizer parameters are updated when a new sample is received. The problem is formulated as minimizing the norm between two subsequent parameter vectors under the constraint of minimal BER/SER. Compared with the existing work that derives the equalizers based on the BER/SER objective functions, our proposed problem model has the following properties: The MSER criterion is reformulated as the constraint of correct symbol detection. It does not need to calculate the BER/SER expressions and hence has a simpler derivation. For different source modulation schemes, our proposed model applies a same objective function, Hence it has a uniform framework for different source modulation schemes.

**KEYWORDS**- Minimum mean squared error (MMSE); Bit Error Rate(BER);Binary Phase Shift Keying(BPSK);Quadrature Amplitude Modulation(QAM);Signal to Noise Ratio(SNR).

## I INTRODUCTION

The Digital communication is becoming more dominant over analog communication. The main factors for this situation are the increasing performance and decreasing cost of digital communication equipment. Communication involves transfer of some information like voice, image or data from a source to a destination with almost in its original form. Then, the sequence is encoded to appropriate symbols and modulated digitally for transferring the information into suitable signal waveforms that can be carried in the channels dedicated for communication. The signal passes the channel and arrives at the receiver. In the receiver of the digital communication signal performs the inverse operation. The received signal is demodulated and the obtained symbols are converted into a binary sequence. The binary sequence is used to form the original information by a source decoder. The transmitter is located at one point in space; the receiver is located at some other point separate from the transmitter. The capacity of direct-sequence code-division multiple-access (DS-SS) systems is limited primarily by the near-far problem. This has motivated considerable effort to develop near-far resistant multiuser detectors. Linear multiuser detectors such as the decorrelating detector and the minimum mean-squared error (MMSE) detector are among the most popular due to a number of advantages. The channel is the physical medium that connects the TX and Rx. The purpose of TX is to convert the message signal produced by the source of information into a form suitable for transmission over a channel. However, as the transmitted signal propagates along the channel, it's distorted due to channel imperfections. The receiver has the task of operating on the received signal so as to reconstruct a recognizable form of the original message signal.

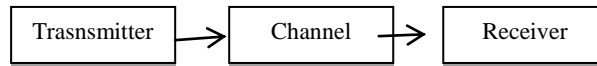


Fig 1 Basic Elements of Communication System.

In existing research, the MSER equalizer is derived based on the bit error rate (BER) or SER objective function, which are for BPSK sources and QAM sources, respectively. The BER/SER objective functions are non-convex and consequently it is difficult to obtain a closed-form solution. Iterative approaches usually are applied. In the gradient descent method is applied to find a local minimum of the BER/SER objective functions. By setting the gradient to zero. Adaptive equalization with the following attributes: it has low complexity; it does not require knowledge of the channel or noise power; it has good convergence properties.

## II. BACK GROUND

### A. BPSK Sources

For BPSK sources, correct symbol detection satisfies the following conditions:

$$\begin{cases} c_k^T r_k > 0, & \text{when } s_{k-D} = 1 \\ c_k^T r_k < 0, & \text{when } s_{k-D} = -1. \end{cases} \quad (1)$$

that the minimum-BER is equivalent to ensuring a correct symbol detection, which is formulated as the following constraint:

$$\text{sgn}(c_k^T r_k) = \begin{cases} 1, & \text{when } s_{k-D} = 1 \\ -1, & \text{when } s_{k-D} = -1. \end{cases} \quad (2)$$

BPSK (also sometimes called PRK, phase reversal keying, or 2PSK) is the simplest form of phase shift keying (PSK). It uses two phases which are separated by 180° and so can also be termed 2-PSK. It does not particularly matter exactly where the constellation points are positioned, and in this figure they are shown on the real axis, at 0° and 180°. This modulation is the most robust of all the PSKs since it takes the highest level of noise or distortion to make the demodulator reach an incorrect decision. It is, however, only able to modulate at 1 bit/symbol and so is unsuitable for high data-rate applications. In the presence of an arbitrary phase-shift introduced by the communications channel, the demodulator is unable to tell which constellation point is which. As a result, the data is often differentially encoded prior to modulation.

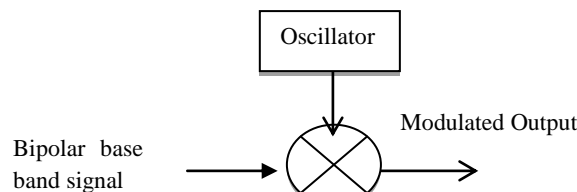


Fig 2. BPSK Modulator.

### B. QAM Sources

The fundamental problem with PSK is that of determining the phase of the transmitted signal and hence decoding the transmitted information. This problem is also known as carrier recovery as an attempt is made to recover the phase of the carrier. When a phase point at, say, 90 deg is selected to reflect the information being transmitted, the phase of the

transmitted carrier is set to 90 deg. However, the phase of the carrier is often changed by the transmission channel with the result that the receiver measures a different phase. This means that unless the receiver knew what the phase change imposed by the channel was it would be unable to determine the encoded information. This problem can be overcome in one of two ways. The first is to measure the phase change imposed by the channel by a variety of means. The receiver can then determine the transmitted phase. This is known as coherent detection. The second is to transmit differences in phase, rather than absolute phase. The receiver then merely compares the previous phase with the current phase and the phase change of the channel is removed. This assumes that any phase change within the channel is relatively slow. This differential system is known as non-coherent transmission. In his paper, Cahn considered both coherent and non-coherent transmission, although for coherent transmission assumed a hypothetical and unrealizable perfect carrier recovery device.

#### C. Particle Swarm Optimization

PSO (Particle Swarm Optimization) no operators inspired by natural evolution are applied to extract a new generation of candidate solutions. Instead of mutation PSO relies on the exchange of information between individuals, called particles, of the population, called swarm. In , each particle adjusts its trajectory towards its own previous best position, and towards the best previous position attained by any member of its neighborhood . In the global variant of PSO, the whole swarm is considered as the neighborhood.

A large inertia weight facilitates exploration while a small one tends to facilitate exploitation. A proper value for the inertia weight  $w$  provides balance between the global and local exploration ability of the swarm, and, thus results in better solutions. Experimental results imply that it is preferable to initially set the inertia to a large value, to promote global exploration of the search space, and gradually decrease it to obtain refined solutions distributed within the search space.

#### D. Impulse Response

The impulse function is defined as an infinitely high, infinitely narrow pulse, with an area of unity. This is, of course, impossible to realize in a physical sense. If the impulse width is much less than the rise time of the filter, the resulting response of the filter will give a reasonable approximation actual impulse response of the filter response. The impulse response of a filter, in the time domain, is proportional to the bandwidth of the filter in the frequency domain. The narrower the impulse, the wider the bandwidth of the filter. The pulse amplitude is equal to  $\omega c/\pi$ , which is also proportional to the filter bandwidth, the height being taller for wider bandwidths. The pulse width is equal to  $2\pi/\omega c$ , which is inversely proportional to bandwidth. It turns out that the product of the amplitude and the bandwidth is a constant. It would be a nontrivial task to calculate the response of a filter without the use of Laplace and Fourier transforms .The impulse response of a filter, in the time domain, is proportional to the bandwidth of the filter in the frequency domain. The narrower the impulse, the wider the bandwidth of the filter. The pulse amplitude is equal to  $\omega c/\pi$ , which is also proportional to the filter bandwidth.

Discrete-time unit impulse and unit step functions

$$\delta[n] = \begin{cases} 0, & n = 0 \\ 1, & n \neq 0 \end{cases}, \quad \text{and} \quad u[n] = \begin{cases} 0, & n < 0 \\ 1, & n \geq 0 \end{cases}.$$

### III. NOISE AND DISTORTION

Noise can be defined as an unwanted signal that interferes with the communication or measurement of another signal. A noise itself is a signal that conveys information regarding the source of the noise. For example, the noise from a car engine conveys information regarding the state of the engine. The sources of noise are many, and vary from audio frequency acoustic noise emanating from moving, vibrating or colliding sources such as revolving machines, moving

vehicles, computer fans, keyboard clicks, wind, rain, etc. to radio-frequency electromagnetic noise that can interfere with the transmission and reception of voice, image and data over the radio-frequency spectrum. When the signal statistics are known, this knowledge can often be used to design suitable detectors. In the situation treated here for known of the signal form that one is reluctant to make unwanted assumptions about it. However, the signal is considered to be deterministic although unknown in detail. The spectral region to which it is approximately confined is, however, known. The noise is assumed to be Gaussian and additive with zero mean. The assumption of a deterministic signal means that the input with signal present is Gaussian but not zero mean.

#### IV. COMPUTER SIMULATION

Computer simulation results are presented to demonstrate the effectiveness of the adaptive Equalizer. The performance difference between an exact minimum SER(EMSER) equalizer is most pronounced when the number of equalizer coefficients is small relative to the severity of the ISI.

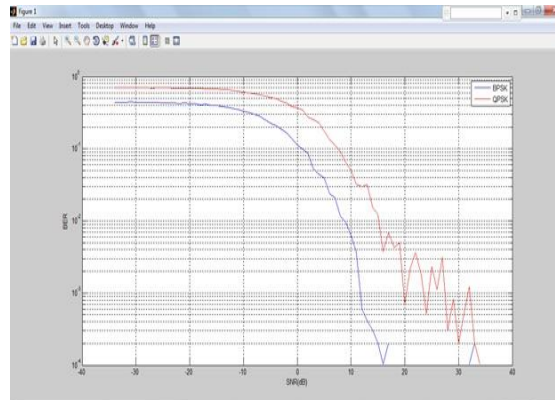


Fig.3. The convergence speed of several different step sizes for a BPSK system,



Fig.4. The convergence properties of the LMS algorithm in a QPSK system

#### V. CONCLUSION

This work was presence in the reducing symbol error rate using BPSK,QPSK modulation schemes and adaptive equalizer. Moreover, by introducing a normalization factor, the proposed MSER equalizer provides better performance in convergence speed and steady-state SER..

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