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COMPARISON OF LINEAR AND NON-LINEAR MULTIUSER DETECTORS WITH QUANTUM CONSTRAINTS

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ABSTRACT

In this paper, a detailed comparison between different multiuser detectors for synchronous code-division multiple-access (CDMA) systems has been analyzed. Here speech signal is being converted into vectors in the form of matrix. Each vector in the matrix indicates the samples obtained from the speech signal. An algorithm has been described as measurement vector orthogonality and imposes an inner product constraint. Unlike the linear minimum mean-squared error (LMMSE) receiver, other multiuser detectors depends only on the signature vectors and does not require knowledge of the received amplitudes or the channel signal-to-noise ratio (SNR). The implementation can be done for linear and non linear detectors. The simulation results provided suggest that in certain cases the Maximum likelihood sequence estimator performs better with lower bit error rate when compared to linear equalizer and Decision feedback equalizer.

***Keywords**: Multiuser Detection, Minimum mean squared error, Signal to Noise ratio, Bit error Rate, Equalizer.*

1. INTRODUCTION

Code Division Multiple Access (CDMA) systems are very attractive for mobile communication applications because of their efficient use of the channel and their allow ness for nonscheduled user transmissions. Direct sequence (DS) CDMA is a method, which enables the users to share the same RF channel to transmit data simultaneously. The DS CDMA transmitter multiplies each user's signal by a distinct code waveform. The detector receives a signal composed of the sum of all users' signals, which overlap in time and frequency. The detector receives a signal composed of the sum of all users' signals, which overlap in time and frequency. In a conventional DS CDMA system, a particular users' signal is detected by correlating the entire received signal with users' code waveform. During the first part of this decade, it was realized that it was possible to implement an adaptive multiuser detector. This multiuser detector operates directly on the received signal, sampled once or several times during each chip period.

Such a receiver is relatively insensitive to errors in the estimates of the propagation delay and other system parameters. Another advantage is that it can be implemented in such a way that only signals that are of interest need to be detected. The adaptive multiuser detector has two major disadvantages. First, it cannot be applied in systems with long codes. Second, in some cases the rate of convergence may be too slow. The other group of detectors is based upon Interference Cancellation(IC). The idea is to cancel the interference generated by users other than the desired user. Lower computation demand and hardware related structures are the major advantages of this strategy. This category includes Serial Interference Cancellation (SIC) [17, 18] and Parallel Interference Cancellation (PIC). One of the most effective PICs comes from the iterative multistage method [19]. The inputs of one particular stage are the estimated bits of the previous stage. After interference cancellation, the new estimations, which should be closer to the transmitted bits, come out to be fed into the next stage.

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The DS/CDMA receivers are divided into Single-User and Multi-User detectors. A single user receiver detects the data of one user at a time whereas a multi-user receiver jointly detects several users information. Single user and multi user receivers are also sometimes called as decentralized and centralized receivers respectively [1]. At the receiver, the aim is to restore the signal, which is corrupted by the channel back to its original form. In its simplest form, the Single-User detector is a matched filter to the desired signal.

Other user's signals are treated as noise (self noise). These self-noise limit the systems capacity and can jam out all communications in the presence of a strong near by signal (Near-Far Problem). The capacity is optimized when all users enter the base station at the same power level forcing the use of power control circuits in the terminal transmitters. In direct-sequence code division multiple-access (DS-CDMA) systems all users concurrently share the same bandwidth. The users are distinguished by assigning to each user a unique code or signature sequence, whose bandwidth is much larger than that of the transmitted information. This code sequence is used to modulate the data stream.

Detectors based on this principle are known as adaptive multi-user detectors .Multi-user receivers have the potential to significantly improve the performance and capacity of a DS/CDMA system. Interference cancellation is one of the approaches for Multi-User Detection (MUD). Multi-user detection deals with the development and application of joint demodulation and interference cancellation techniques for improved detection of a desired set of digital signals.

2. LITERATURE REVIEW

In [1], an overview of the most common strategies of multiuser detection can be found. Extensive references to relevant research work are found in the book of Verdu [2], where each chapter is ended with bibliographical notes. In 1986 Verdu proposed the novel idea that detection of CDMA signals should exploit the structure inherent in the MAI and not just treating it as noise [2]. With this notion, the conventional matched filter is no longer the optimal detector, and there is a class of multiuser detectors, which are able to reduce MAI and hence, lead to better performance. The simplest scheme to demodulate CDMA signal is to use the conventional matched filter detector, which is optimal in AWGN noise but is sub-optimal because the MAI does not necessarily resemble white Gaussian noise [2].

The received signal is passed through a bank of matched filters attached in Rake configuration that coherently demodulates and dispreads each of the received paths [3]. The problem of this receiver arises from the fact that, even if the powers of all users are equal, some cross correlation among signals might still have high values due to different path delays. Therefore even by adjusting the power level using fast power control and selecting codes with low cross correlations, the performance of the matched filter receiver is limited and so is the capacity since, to maintain acceptable interference limits, the number of users have to be reduced.

P.Rapajic and B.S.Vucetic [4] presented a receiver structure in which the bank of matched filters is replaced by an adaptive fractionally spaced LMS filter [5]. In [6] a bank of LMS filters replaces the user's bank of matched filters attached to the RAKE. In this case channel estimates are also required making the existence of training sequences again necessary.

In [6] it is stated that the optimum detection problem may be solved using Viterbi algorithm. It is also asserted that optimal detection in the case of an asynchronous system requires knowledge of the entire transmitted sequence of each user. Among the DS-CDMA detectors utilizing knowledge of the interferers the first is the detector proposed by Schneider in [7]. The optimal multiuser detector for CDMA systems using Viterbi's algorithm and assuming a perfect knowledge of the channel, is proposed by Verdu in the mid of 1980's in [8-10]. The optimal detector developed in [9] by Verdu presents the optimal solution for the asynchronous case.

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3. IMPLEMENTATION OF MULTIUSER DETECTORS

In a communication system, the transmitter sends the information over an RF channel. The channel distorts the transmitted signal before it reaches the receiver. The receiver task is to figure out what signal was transmitted Equalization compensates for intersymbol interference created by multipath within time dispersive channel. An equalizer within a receiver compensates for the average range of expected channel amplitude and delay characteristics. The analog device is processed by the decision making device in the receiver. The decision maker determines the value of digital data bit being received and applies a slicing or thresholding operation.

3.1 LINEAR EQUALIZER

There is no feedback path for linear equalizer. The current and the past values of the received signal are linearly weighted by equalizer coefficients and summed to produce the output

$$C(z) = \sum_k c_k z^{-k} \tag{1}$$

The ISI can be completely removed, without taking in consideration the resultant noise enhancement Zero forcing equalizer.

$$C(z) = h^{-1}(z) \tag{2}$$

3.2 ZERO-FORCING EQUALIZER

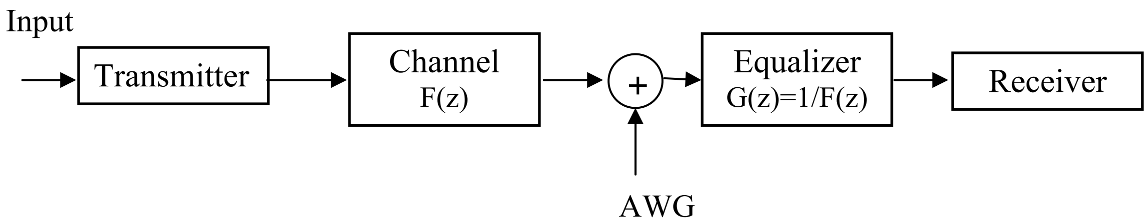

Fig1 Block diagram of Zero Forcing Equalizer

As the name implies, it forces ISI to become zero for every symbol decision. A zero-forcing equalizer enhances noise and results in performance degradation.

3.3 MINIMUM MEAN SQUARE ERROR-LINEAR EQUALIZER (MMSE-LE)

MMSE-LE minimizes the error between the received symbol and the transmitted symbol without enhancing the noise. Although MMSE-LE performs better than ZFE, its performance is not enough for channels with severe ISI. An obvious choice for channels with severe ISI is a non-linear equalizer. From the point-of-view of minimizing error probability, it is power. The MSE criterion attempts to minimize the total error between the slice input and the transmitted data symbol

$$C(z)=1/ (h(z)+2N_0 / \sigma^2_A) \tag{3}$$

3.5 MAXIMUM LIKELIHOOD SEQUENCE EQUALIZER (MLSE)

Maximum Likelihood Sequence Equalizer (MLSE) gives optimum performance. It tests all possible data sequences and chooses the data with the maximum probability as the output. Generally, the Viterbi algorithm provides a solution to the problem with MLSE of a finite-state, discrete-time Markov process. However, the computational complexity of an MLSE increases with channel spread and signal constellation size. The number of states of the Viterbi decoder is expressed as ML, where M is the number of symbols in the constellation, and L is the channel-spread length –1.

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3.6 DECISION FEEDBACK EQUALIZER (DFE)

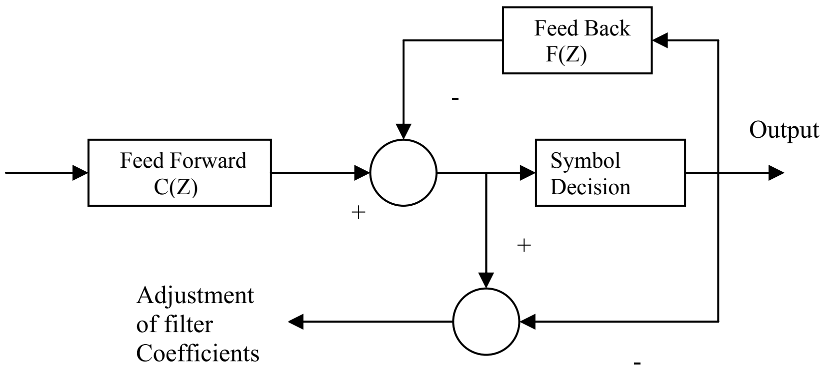

Fig 2 Simple Block diagram of DFE

Simple nonlinear equalizer which is particularly useful for channel with severe amplitude distortion. DFE uses decision feedback to cancel the interference from symbols which have already have been detected. The basic idea is that if the values of the symbols already detected are known (past decisions are assumed correct), then the ISI contributed by these symbols can be canceled exactly.

The smoothing lag of this DFE is infinite, which is equivalent to allowing a non-causal feed forward filter. The multivariable version consists of a channel with two inputs and two outputs and for a more general multivariable channel with equal numbers of inputs and outputs. The optimum realizable DFE, which has causal feed forward and feedback filters. The decisions on the symbols are made after a finite delay. The FIR DFE, where both feed forward filters and feedback filters are transversal filters. The degrees of the feed forward and feedback filter are design variables. A Zero Forcing (ZF) equalizer may provide adequate performance when the noise can be neglected, but at low to moderate signal-to-noise ratios the performance of an MMSE equalizer will be superior. Therefore, an MMSE equalizer is often preferred for practical implementation.

$$Q(k) = \sum_{I=-(N-1)}^{0} C_i Y(K-i) - \sum_{I=1}^{M} f_i \hat{x}(k-i) \tag{4}$$

One sub-optimum multiuser receiver on which extensive research has been done is the linear Minimum Mean Square Error (MMSE) receiver which has a linear complexity, and adaptive techniques (e.g., LMS, RLS algorithms) to approach the MMSE solution are suitable for implementation [7]. The application of nonlinear receivers for interference rejection in DS-CDMA systems has been an active area of recent research. In this project, we focus on the performance of a Decision Feedback Equalizer (DFE) for multiuser detection in DS-CDMA systems. The DFE has a nonlinear structure which consists of two parts: a Feed Forward Filter (FFF) which operates on the outputs of the chip matched filter, and a Feed Back Filter (FBF) which operates on the past decisions of the desired users data as well as those of the interfering users data.

We evaluate the performance of the proposed DFE based receiver on both AWGN as well as Raleigh fading channels in a near-far scenario and compare it with that of linear adaptive receivers which have only feed forward filters. The proposed receiver structure is shown to exhibit good near-far resistance and offer significant performance advantage over linear adaptive receivers. The equalization of orthogonal multi pulse signals has been proposed [2,3] for the specific case of PPM.

In [2] a Zero-Forcing (ZF) Decision Feedback Equalizer (DFE) is proposed that employs an Infinite Impulse Response (IIR) feed forward filter. In [3] the ZF decision feedback equalizer is derived under the following assumptions: the channel is monic and minimum phase, the additive

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noise is ignored (i.e. since it is a ZF equalizer), and the feedback portion of the equalizer is as long as the channel (and possibly infinite).we propose a minimum mean-squared error (MMSE) DFE for orthogonally modulated signals, and we show the design equations and performance of the proposed structure.

We permit nonmonic and non-minimum phase channels, which accommodate noise from any stationary random process, and it permits the length of the FIR feed forward and feedback portions of the equalizer to be the design parameters. We then make several modifications to the equalization structure, thereby permitting a computational savings and the use of stochastic gradient decent techniques for determining the MMSE equalizer tap values. Finally, we include simulation results which demonstrate the performance of the proposed equalizer.

4. FRONT END MODULE (Generation of feature vectors)

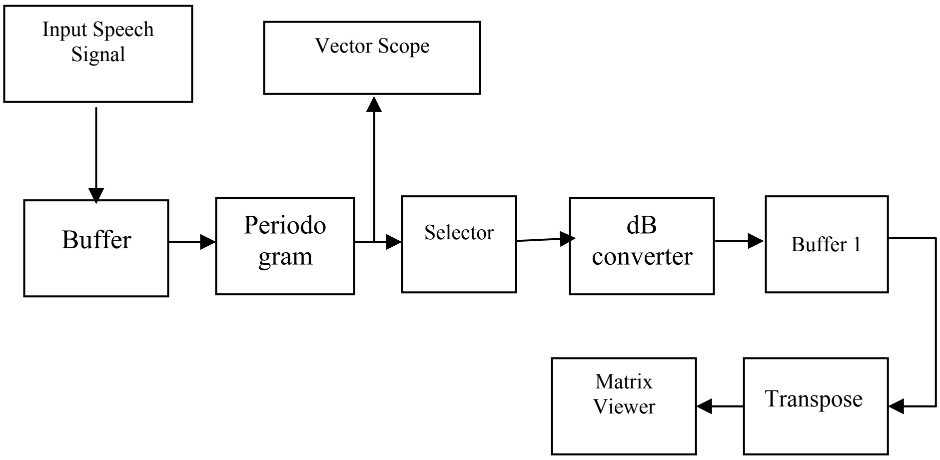

Fig 3 Block diagram of front end module

Spectral analysis is the most important module in front-end processing. There are several ways to extract spectral information of speech. When the audio file contains two channels (stereo), the block's output is an M-by-2 matrix containing one frame (M consecutive samples) of audio data from each of the two channels. When the audio file contains a single channel (mono), the block's output is an M-by-1 matrix containing one frame (M consecutive samples) of mono audio data.

At the start of the simulation, the audio device begins writing the input data to a (hardware) buffer with a capacity of T_b seconds. The Wave Device block immediately begins pulling the earliest samples off the buffer (first in, first out) and collecting them in length-M frames for output. As the audio device continues to append inputs to the bottom of the buffer, the From Wave Device block continues to pull inputs off the top of the buffer at the best possible rate. Here scalar samples are converted to frames output.

Using the Periodogram block nonparametric estimate of the power spectrum of the speech signal is computed. With the use of transpose the vector input signals are treated as (Mx1) matrices as output. For the specified inner product structure another constraint that can be applied for consistency is the least square inner product constraint. Here the obtained optimal vectors must be orthogonal and orthonormal. The optimal vectors that are used to minimize the sum of the mean squared norms of the errors between constructed and the given vectors. For that we go for singular value decomposition and other set transformations also satisfies the condition that the inner product of the resultant matrices will be a identity matrix ,the probability value must be one for all type of vectors.

The important elements of quantum constraints are that the measurement vectors are constrained to be orthonormal and orthogonal. For choosing a set of measurement vectors that

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“best” represent the signals of interest and have a specified inner product structure. Specifically, measurement vectors q_i are constructed with a given inner product structure that are closest in a Least Square (LS) sense to a given set of vectors s_i, so that the vectors q_i are chosen to minimize the sum of the squared norms of the error vectors $e_i = q_i - s_i$. These techniques are referred to as LS inner product shaping has potential applications to a variety of problems.

5. RESULTS AND DISCUSSION

The analysis shows the BER performance of several types of equalizers in a static channel with a null in the pass band. The script constructs and implements a linear equalizer and a decision feedback equalizer (DFE). It also initializes and invokes maximum likelihood sequence estimation (MLSE) equalizer. The MLSE equalizer is first invoked with perfect channel knowledge, then with a straightforward but imperfect channel estimation technique.

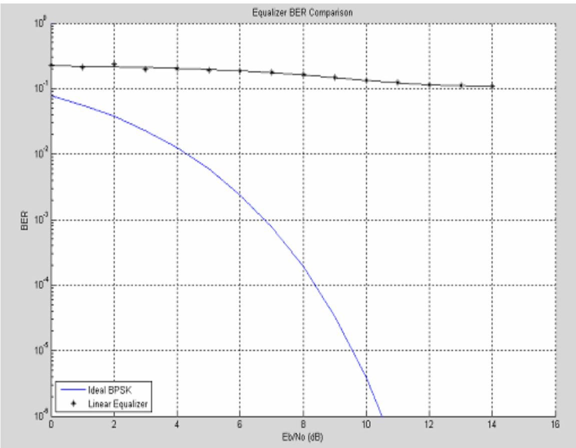

Fig 4 (a) BERVs Eb/No for linear Equalizer

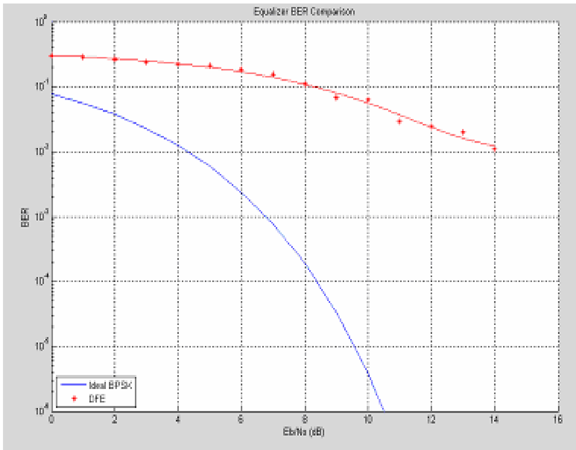

Fig 4 (b) BER Vs Eb/No for Decision feed back Equalizer

As the simulation progresses, it updates a BER plot for comparative analysis between the equalization methods. It shows the relative burstiness of the errors, indicating that at low BERs, both the MLSE algorithm and the DFE algorithm suffer from error bursts. In particular, the DFE error performance is burstier with detected bits fed back than with correct bits fed back. Finally, during the "imperfect" MLSE portion of the simulation, it shows and dynamically updates the estimated channel response. By changing such parameters as the channel impulse response, the number of equalizer tap weights, the recursive least squares (RLS) forgetting factor, the least mean square (LMS) step size, the MLSE trace back length, the error in estimated channel length, and the maximum number of errors collected at each Eb/No value.

For linear and DFE equalizers set the parameter values. We Use the recursive least squares (RLS) algorithm for the first block of data to ensure rapid tap convergence. For the linear equalizer the plot of Eb/No and BER, each data block is performed. Note that as the Eb/No increases, the bit error rate progressively null. This highlights the fact that a linear equalizer must have many more taps to adequately equalize a channel with a deep null. Note also that the errors occur with small inter-error intervals, which is to be expected at such a high error rate.

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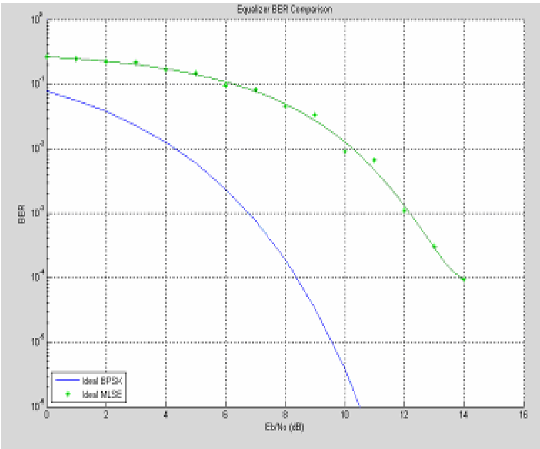

Fig 5 (a) BER Vs Eb/No for MLSE

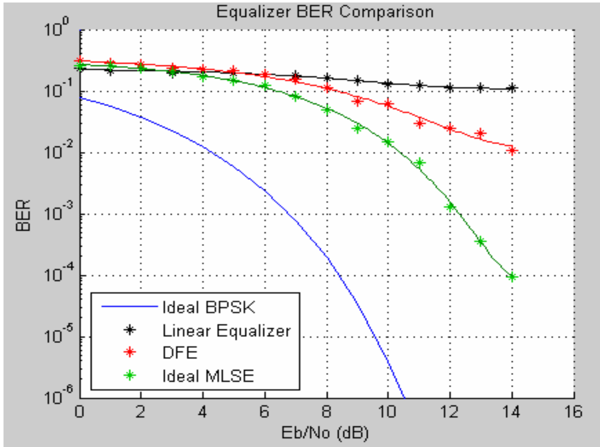

Fig 5 (b) Equalizer BER Comparison

Eb/N_0	Linear equalizer	Decision feedback filter	MLSE	Eb/N_0	Linear equalizer	Decision feedback filter	MLSE
0	0.2302	0.2733	0.2586	8	0.1652	0.1126	0.0287
1	0.2110	0.2910	0.2408	9	0.1465	0.0857	0.0169
2	0.2316	0.2547	0.2378	10	0.1318	0.0818	0.0089
3	0.1947	0.2479	0.1982	11	0.1219	0.0264	0.0047
4	0.2001	0.2278	0.1858	12	0.1131	0.0264	0.0022
5	0.1928	0.2062	0.1699	13	0.1101	0.0298	0.0004
6	0.1883	0.1744	0.1105	14	0.1096	0.0068	0.0000
7	0.1751	0.1396	0.0599				

Table 1 Comparison of Linear, Decision feed back and MLSE equalizer for BER performance Vs Eb/N_0

Next the plot decision feedback equalizer can be made between Eb/No and BER for each data block. Note that the DFE is much better able to mitigate the channel null than the linear equalizer, as shown in the BER plot. The plotted BER points at a given Eb/No value are updated every data block, so they move up or down depending on the number of errors collected in that block. For MLSE equalizer with a perfect channel estimate, the plot between BER and Eb/No for each data block. Note that the errors occur in an extremely bursty fashion. Observe, particularly at low BER's, that the overwhelming percentage of errors occur with an inter-error interval of one or two bits.

CONCLUSION

In this paper we develop methods that construct an optimal set of vectors with a specified inner product structure from a given set of vectors optimal vectors are chosen to minimize the sum of the squared norms of the errors between the constructed vectors and the given vectors. Finally we made ca comparison between different multiuser detectors. First, we showed that the detector can be interpreted as a linear equalizer followed by Decision feedback equalizer that compensates for better bit error rate enhancement without reintroducing too much MAI. Second, we demonstrated Maximum likelihood detector which made as a better performance than the above non linear detector. Corresponding curves for the linear equalizer, Decision feedback equalizer, Maximum likelihood detector for the given speech input signal and are plotted for comparisons. The simulation result shows better performance for non linear detectors than the linear detectors.

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