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Abstract—Orthogonal Frequency Division Multiplexing (OFDM) is gaining popularity for high data rate communication systems. We propose the use of adaptive beamforming for interference rejection in OFDM systems, due to its advantages over equalization. An adaptive beamformer uses the concept of spatial filtering to direct the antenna beam towards the desired signal/transmitter and place a null towards the interfering signal. The Fast Fourier Transform (FFT) at the receiver end of an OFDM system enables the use of frequency domain beamforming to reduce narrow band interference individually across all the subcarriers. We implemented both decision directed and blind algorithms. Use of these algorithms reduced the Bit Error Rate (BER) to a great extent. The performance analysis for Least Mean Square (LMS) algorithm, Recursive Least Squares (RLS) and Constant Modulus Algorithm (CMA) for hundred OFDM symbols and five hundred and twelve subcarriers, four hundred and is provided.

Index Terms— Adaptive Algorithms, Antenna Array, Frequency Domain Beamforming , OFDM.

I. INTRODUCTION

Adaptive arrays are currently the subject of extensive investigation as a means for reducing the vulnerability of the reception of desired signals to the presence of these interference signals in radar, sonar, seismic, and communication systems [1]. They provide an efficient means for minimizing channel interferences by directing the antenna beam towards the desired signal/transmitter and place a null towards the interfering signal.

Orthogonal Frequency Division Multiplexing (OFDM) is a potential candidate for future high-bit-rate wireless communication systems as is less susceptible to intersymbol interference (ISI) introduced in the multipath environment [2]. OFDM is a multicarrier technique that sends information on a number of overlapping and orthogonal subcarriers by dividing the total signal bandwidth. A Guard time longer than the channel delay spread is introduced in each OFDM symbol to mitigate ISI. However symbols still experience interference by their replicas termed as self-interference. Under the assumptions of narrowband model phase shift is introduced in each multipath component. One solution for removal of this phase shift is equalization that requires channel estimation which is difficult when the power of the interfering signal is higher than the desired signal. Many blind algorithms [3] and a blind equalization criterion [4] have been developed for removing this phase shift. The transmitted signal has some cyclostationary properties that can be used in correlation matching techniques as in [5] and [6]. The difficulty in

implementing these methods arises from null side carriers. An approach that can work with transmitters inserting guard time in the symbols is presented in [7]. The drawback of this approach is that requires that the guard time length be equal to the block size. This requires a large overhead. An improved subspace algorithm is provided in [8] but it does not provide accuracy for channel estimation in the frequency domain.

The approach we suggest bypasses equalization and employ an antenna array with an adaptive beamforming algorithm as phase shift can be removed by employing an adaptive antenna array.

The rest of the paper consists of following sections. Section 2, the system model, describes OFDM symbol generation, multipath channel model and our proposed receiver design. Section 3 briefly describes adaptive algorithms analyzed in the paper. Section 4 discusses performance analysis of the adaptive algorithms.

II. SYSTEM MODEL

An OFDM symbol generated by an N -subcarrier OFDM system consists of N samples. The m -th sample can be represented by (1) [9]

$$x_m = \sum_{n=0}^{N-1} X_n \exp \{j2\pi mn/N\} \quad (1)$$

Where X_n is the data symbol transmitted on the n th subcarrier. This is similar to taking Inverse Discrete Fourier Transform (IDFT) which can be easily implemented using Inverse Fast Fourier Transform (IFFT). To reduce the ISI, a guard time longer than the delay spread is inserted cyclically at the beginning of each OFDM symbol before transmission, maintaining orthogonality among the subcarriers. It is removed at the receiver before the FFT operation. As long as guard time is greater than delay spread, only a different phase shift for each subcarrier is introduced and the orthogonality among subcarriers is maintained. If G is the guard time then m -th sample after guard time insertion is expressed as [9]

$$x_m = \left(\sum_{n=0}^{N-1} X_n \exp \{j*(2\pi mn)/N\} \right)_{(m+N-G)} \quad (2)$$

After being translated to a high carrier frequency signal is transmitted through a multipath channel.

Using the narrowband model assumption, the m -th sample received at the k -th antenna element can be written as [9]

$$r_{m,k} = \sum_{l=0}^{L-1} h_{m,l} x_m \exp \{-j(2\pi/\lambda(k-1)d \sin\Theta)\} + n_{m,k} \quad (3)$$

Where

- d is spacing between the antenna elements,
 λ is the carrier wavelength,
 Θ is the angle of arrival of signal at the antenna element with respect to array normal,
 $h_{m,l}$ is the complex random variable that represents channel's impulse response for the l -th path of the channel at time m ,
 $n_{m,k}$ is AWGN at the k -th antenna element at time m .

After the removal of Guard time FFT is implemented to demodulate the received symbols. The demodulated symbol, on the n -th subcarrier at the output of FFT, at the k -th antenna element can be written as [10]

$$Y_{n,k} = \sum_{m=0}^{N-1} \sum_{l=0}^{L-1} X_m H_l(n-m) \exp\{-j(2\pi m l / N + 2\pi / \lambda (k-1) d \sin \Theta)\} + N_{n,k} \quad (4)$$

$$= \sum_{l=0}^{L-1} H_l(n) \exp\{-j(2\pi n l / N + 2\pi / \lambda (k-1) d \sin \Theta)\} X_n + N_{n,k}$$

$$+ \sum_{m=0, m \neq n}^{N-1} \sum_{l=0}^{L-1} X_m H_l(n-m) \exp\{-j(2\pi m l / N + 2\pi / \lambda (k-1) d \sin \Theta)\} \\ = \alpha_{n,k} + \beta_{n,k} + N_{n,k} \quad (5)$$

Where

$N_{n,k}$ is the AWGN on the n -th subcarrier at the k -th antenna element,

$\alpha_{n,k}$ is the multiplicative distortion caused by the channel at the desired subcarrier at the k -th antenna element,

$\beta_{n,k}$ is the inter channel interference (ICI),

$H_l(n-m)$ is the FFT of a time-variant multipath channel given by:-

$$H_l(n-m) = (1/N) \left[\sum_{k=0}^{N-1} h_{m,l} \exp\{-j(2\pi k(n-m)/N)\} \right] \quad (6)$$

FFT taken at the receiver side allows frequency domain beamformer to be implemented separately for individual subcarriers. Having individual beamformers to process its own subcarriers is an advantage for suppressing the narrowband interference. Narrowband interference corrupts only a portion of the signal bandwidth. Having multiple beamformers across the signal bandwidth provides flexibility so that individual beamformers adjust their weights to adapt different interference patterns experienced by different subcarriers. These weights are updated iteratively using adaptive algorithms which are commonly classified in two types i.e. decision directed algorithms and blind algorithms. Decision directed algorithms use training symbols for error estimation. Blind algorithms on the other hand, do not require any prior knowledge of the received signal. In this paper we provide a performance analysis of these algorithms for an OFDM system. To implement decision directed algorithms, we employed Least Mean Square (LMS) algorithm and Recursive Least Squares (RLS) algorithm and for the case of Blind

algorithm, Constant Modulus Algorithm (CMA) is used. The output of the filter is calculated by the equation [11]:

$$y(n) = w^T(n)x(n) \quad (7)$$

Where

$y(n)$ is the output of the filter

$x(n)$ is the input sequence to the filter

$w(n)$ is the vector containing weights of the filter

III. ADAPTIVE ALGORITHMS

A. Least Mean Square (LMS) Algorithm

The LMS algorithm is a method of stochastically implementing the steepest descent algorithm[5] Successive corrections to the weight vector in the direction of the negative of the gradient vector eventually lead to the Minimum Mean Square Error (MMSE), at which point the weight vector assumes its optimum value. The equations employed are [11]:

$$w(n+1) = w(n) + 2\mu * e(n) * x(n) \quad (8)$$

Where

$e(n)$ is the error estimate given by

$$e(n) = d(n) - y(n) \quad (9)$$

μ is the step size, which controls the speed of convergence. Mean Square Error (MSE) is increased with increase in step size and is decreased according to decrease in the step size [12]. A plot of MSE vs. different values of μ is shown in fig. 3. To ensure convergence of the weight vector, the range of step size is given by [11]:

$$0 < \mu < 1 / \lambda_{\max} \quad (10)$$

B. Recursive Least Square (RLS) Algorithm

RLS is a deterministic algorithm in which the performance index is the sum of weighted error squares for the given data. The tap weight vector update equation is, [11]

$$w(n) = w(n-1) + k(n) * e_{n-1}(n) \quad (11)$$

Where

$e_{n-1}(n)$ is error estimate given by,

$$e_{n-1}(n) = d(n) - y_{n-1}(n) \quad (12)$$

$k(n)$ is gain vector given by,

$$k(n) = u(n) / (\lambda + x^T(n) * u(n)) \quad (13)$$

Where

$$u(n) = \psi_{\lambda}^{-1}(n-1)x(n) \quad (14)$$

where is updated through the equation

$$\psi_{\lambda}^{-1}(n) = \lambda^{-1}(\psi_{\lambda}^{-1}(n-1).k(n) * [x^T(n) * \psi_{\lambda}^{-1}(n-1)]) \quad (15)$$

Where λ is known as forgetting factor that determines the emphasis put by the algorithm on the previous samples of the received data [11]

C. Constant Modulus Algorithm (CMA)

The constant modulus algorithm is a blind adaptive algorithm proposed by Goddard [13] and by Treichler and Agee [14]. That is, it requires no previous knowledge of the desired signal. Instead it exploits the constant or nearly constant amplitude properties of most modulation formats used in wireless communication. the error estimate is given by:

$$e(n) = y(n) / |y(n)| - y(n) \quad (16)$$

When the CMA algorithm converges, it converges to the optimal solution, but convergence of this algorithm is not guaranteed [15]

IV. SIMULATION DISCUSSION

A bit stream generated by the source is modulated using shifted Quadrature Phase Shift Keying (QPSK). Then these bits are converted from serial-to-parallel and virtual carriers are added before subcarrier modulation, implemented through IFFT. Finally the baseband symbol is modulated using a high frequency carrier. The OFDM symbols are transmitted through a two ray channel with Additive White Gaussian Noise (AWGN), frequency response of the channel is given in Figure 1a and Figure 1b. Impulse response showing the delay spread of the channel is given in Figure 2. The constellation diagram of the received symbols through this channel is shown in figure 4. When at the receiver end individual frequency domain beamformers for all subcarriers are placed noise in the constellation diagram is completely eliminated once the adaptive algorithm has converged as shown in fig. 5. The beamformers are implemented using LMS, RLS and CMA algorithm. A plot of mean square error versus iterations is shown in fig. 6, 7, 8 for LMS, RLS and CMA respectively for 512 subcarriers. It is evident from these figures that convergence for LMS is very slow whereas RLS converges faster but it is computationally more complex as it involves matrix inversion. The performance of these algorithms improves as the number of symbols is increased. A plot of bit error rate and the number of symbols for LMS is given in fig. 9. The use of separate beamformer for each subcarrier makes the system complex; however for a less frequency selective channel it is possible to use one beamformer for more than one subcarrier. For the 2-ray channel the BER remains same when one beamformer is used for 1, 4 or 8 subcarriers but for 16 subcarriers the BER rises tremendously.

V. SIMULATION RESULTS

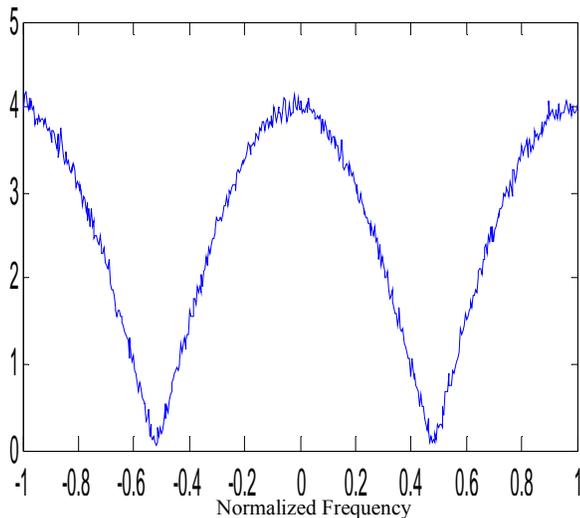


Figure 1a magnitude plot of channel frequency response

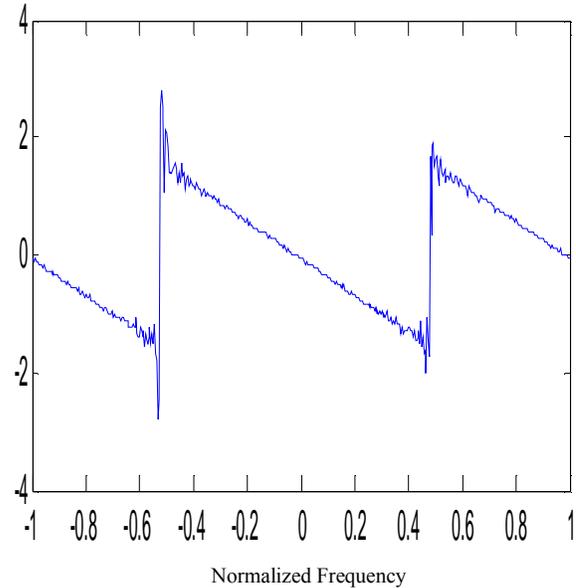


Figure 1b, phase plot of channel frequency response

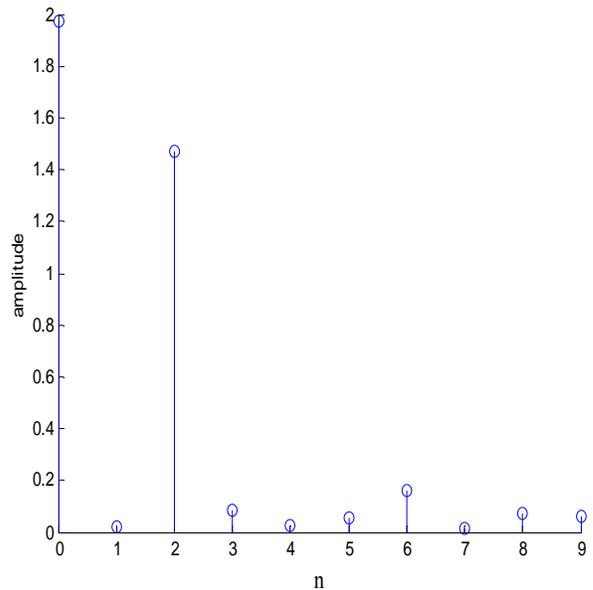


Figure 2, channel impulse response

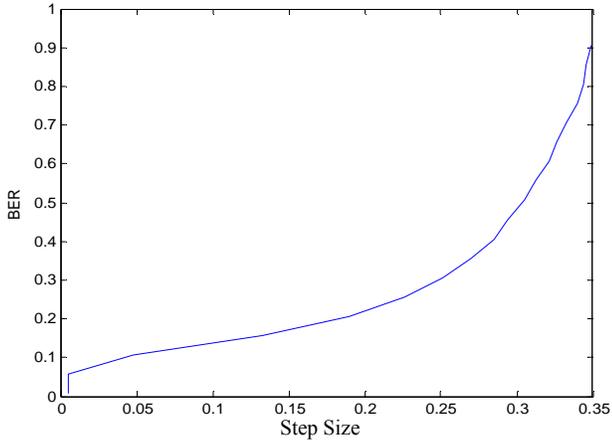


Fig. 3, Plot of BER vs. μ

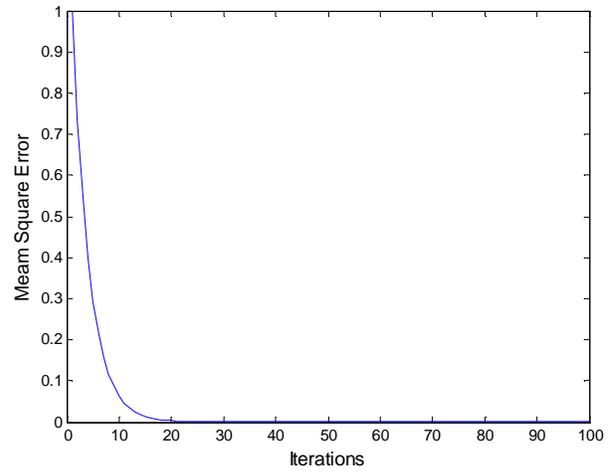


Fig. 6, plot of mean square error vs. number of iterations for LMS

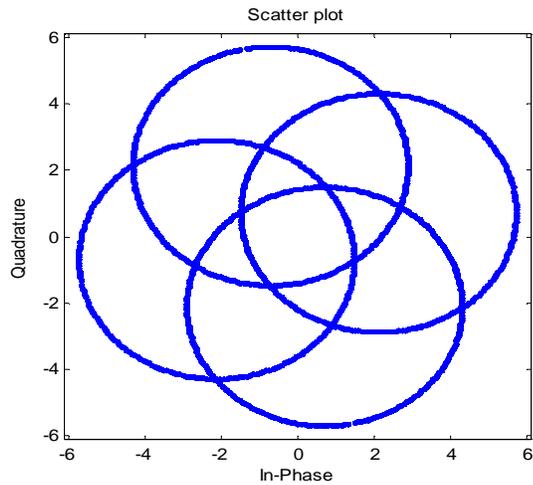


Fig. 4, Constellation Diagram of the received PSK symbol without adaptive algorithm

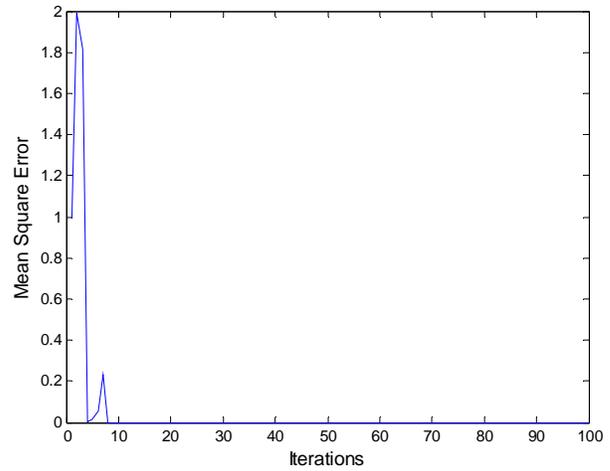


Fig. 7, plot of mean square error vs. number of iterations for RLS

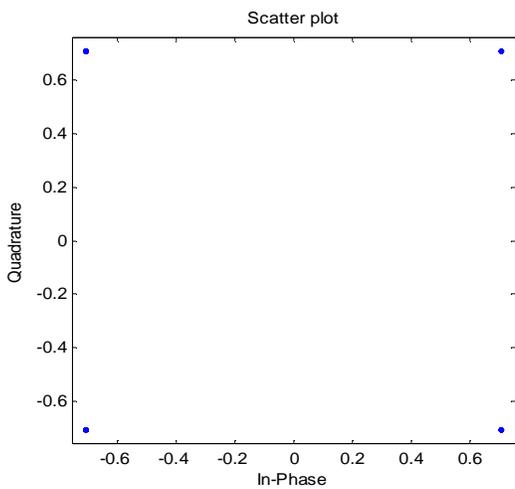


Fig. 5, Constellation Diagram of the received PSK symbol after adaptive algorithm has converged

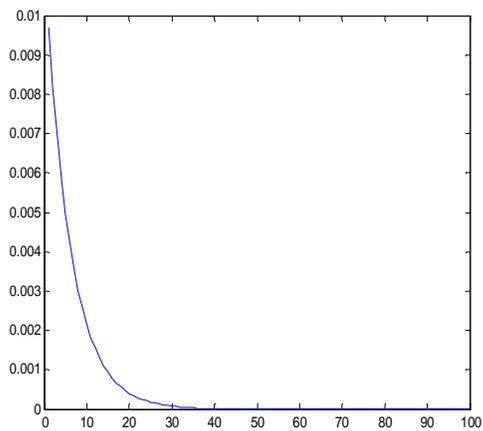


Fig. 8, plot of mean square error vs. number of iterations for CMA

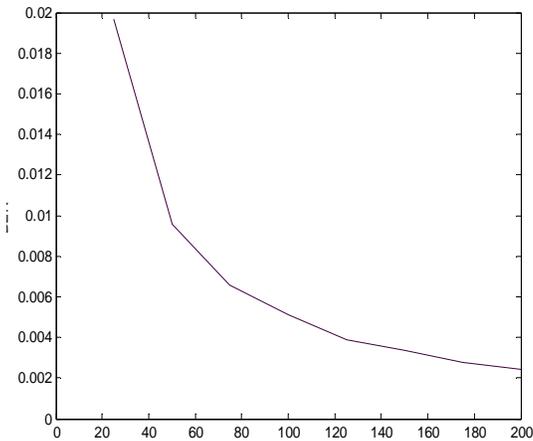


Fig. 9, plot bit error rate vs. number of symbols

VI. CONCLUSION

The use of adaptive beamforming technique can reduce the effects of signal distortion introduced by multipath environment, thus reducing bit error rate of the received signal. This technique can also be used for a time-varying multipath channel. As it does not require equalization, channel estimation is avoided. The decision directed algorithms (LMS & RLS) require training symbols for updating the weight vector of the beamformer, which reduces the bandwidth efficiency of the system below maximum. The convergence of blind algorithm i.e. CMA is very slow and not guaranteed.

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