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Performance Comparison of Decorrelating Rake Receiver and MMSE Detector

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***Abstract:-* In a wireless communication system, the signal can reach the receiver via multiple distinct pathways. In each path, the signal can be blocked, reflected, diffracted and refracted. The signal of this many routes reach receivers faded. The Rake Receiver is a radio, whose goal is to try to minimize the effects of the signal fading due to multipath. In fact, we can understand a set of Rake Receiver sub-radios, each lagged slightly, and to allow the individual components of the multipath can be tuned properly. In order to reduce multi access interference (MAI), a de-correlating matched filter is implemented to improve system performance. A matched filter is used in the first stage followed by post processing stage to reduce multi user interference. Channel estimation and symbol estimation is also done using this method. Finally the performance is evaluated by calculating bit error rate (BER) and plotting it against various SNR values for different spreading factors.**

Keywords-- Rake Receiver, MMSE detector, MAI, Equalizer, Matched filter, Equalization, Adaptive equalizer, LMS.

I. INTRODUCTION

As demand for wireless communications continues to grow, third-generation cellular communications systems are being standardized to provide flexible voice and data services. Standardization bodies around the world are developing systems based on direct-sequence code-division multiple-access (DS-CDMA). In North America, the second generation DS-CDMA standard IS-95 is being used as a basis for a third-generation system (IS-2000) with wider bandwidth. In Japan and Europe, a third-generation wideband CDMA (WCDMA) [1] system, is also being developed. Currently, there is significant effort to harmonize and merge these systems into a common, global third generation CDMA standard.

From a receiver perspective, a wider bandwidth usually corresponds to a smaller chip period, increasing multipath resolution. In the downlink, where user signals from the same base station pass through the same dispersive channel, increased multipath resolution results in interference with significant spectral distortion. Also, the increased multipath resolution leads to loss of orthogonality to interferers within the cell in systems using orthogonal spreading codes.

Rake receivers are widely used as receiver in case of WCDMA systems. When the signal bandwidth is large and multipath components are delayed by a chip period, received signal can be treated as multiple copies of same signal, and can be combined constructively in rake receiver. This rake receiver overcomes the effect of multipath fading as shown in Fig 1. When the spreading codes that are used for the spreading of the signal have good auto-correlation and cross-correlation property, i.e. auto-correlation is high and cross-correlation is as low as zero, rake receiver is able to overcome the effect of the multi-access interference (MAI) [2]. But practically in high data rate transmission through frequency selective channels, the signal loss orthogonality and hence, rake receiver performs poorly. The loss of the orthogonality has an adverse effect on both channel estimation and symbol detection and thus the performance degradation occurs when the system is heavily loaded i.e. when the number of users increases.

The basic idea of a RAKE receiver was first proposed by Price and Green. These fellows also filed the RAKE receiver patent in 1956. [1]

RAKE RECEIVER

Due to reflections from obstacles a radio channel can consist of many copies of originally transmitted signals having different amplitudes, phases, and delays. If the signal components arrive more than duration of one chip apart from each other, a RAKE receiver can be used to resolve and combine them. The RAKE receiver uses a multipath diversity principle. It is like a rake that rakes the energy from the multipath propagated signal components.

In this paper channel estimation [8] and symbol detection is performed using de-correlating rake receiver. The received symbol at the de-correlating matched filter front end is passed through the signal space of each user. After that the channel and data sequence can be estimated independent of other users by least square using singular value decomposition. The de-correlating matched filter does not depend on the channel coefficient. Here the scheme imposes no conditions on channel parameters and is thus capable of dealing with rapid multipath fading.

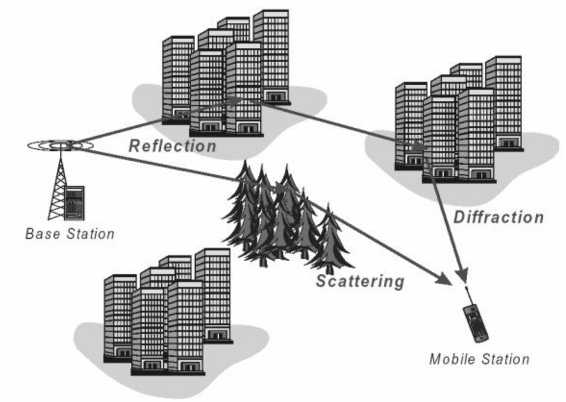

Fig 1. Propagation mechanisms

Following the introduction, the rest of the paper is organized as follows. Section II represents the system model of the receiver system. Section III describes the adaptive equalization technique that is used and section IV presents simulation results i.e. the performance of the work. Finally section V gives the concluding remark.

II. SYSTEM MODEL

A. matched filter

Matched filter bank is usually the first stage in the base band signal detection. Almost all modern multi-user detection techniques deal with the output of the matched filter bank and the cross-correlation information of all users in the system. The single user matched filter consists of 'K' filter banks which are used to receive signal and extract the required message signal through correlation. In matched filter was designed for orthogonal signature waveforms, which correlates the received waveform with the suitably delayed version of the spreading code. It does not cancel the effect of interference (MAI) from other users.

The main drawback of Matched filter in CDMA system is the Multiple Access Interference (MAI) signal. It is high at the output of the Matched filter. Multiple Access Interference (MAI) is a type of interference in multiple cellular users. Multiple access interference (MAI) is an interference which limits the capacity and performance of CDMA systems. This interference is the result of the random time offsets between signals, which make it impossible to design the code waveforms to be completely orthogonal.

The output of the each matched filter can be represent as $y_1[t], y_2[t], \ldots, y_k[t]$.Here $y_1[t]$ is the output of the matched filter and so on.

Received signal at base band is given by

$$y(t) = \sum_{k=1}^{n} A_k b_k S_k(t) + \sigma n(t) \qquad (1)$$

Where,

$S_k(t)$ = deterministic signature waveform assigned to the kth user, normalized so as to have unit energy

A_k = received gain of the linear time invariant channel for user k.

b_k = bit transmitted by the kth user values must be either1's and -1's

n(t) = white Gaussian noise with unit power spectral density. The sampled output of the matched filter for kth user is

σ^2= Covariance matrix

$$Y_k = \int_0^T Y_t . S_k(t) . dt \ldots\ldots\ldots\ldots \quad (2)$$

In Matrix representation, output of the Matched filter is given by

$$Y = R.A.b + n \ldots\ldots\ldots\ldots \quad (3)$$

Where,

R = the normalized cross correlation matrix whose diagonal elements are equal to 1 and whose (i,j) elements is equal to the cross-correlation r_{ij}.

A= diag$\{A_1, \ldots, A_k\}$

y= $[y_1 \ldots y_k]^T$.

b= $[b_1 \ldots b_k]^T$

n = a Gaussian random vector with zero mean

B. Decorrelator

The de-correlator is a linear detector which applies a linear transformation to the matched filter output to reduce the effect of multiple access interference (MAI), hence is near-far resistant. The transformation R^{-1} is applied in a de-correlator which eliminates the MAI signal. The detector that can cancel the MAI signals completely is the de-correlator detector.

The output of the Decorrelator Rake Receiver [6] is

$$R^{-1}y = R^{-1}RAb+n$$

$$R^{-1}y = Ab+n \qquad (4)$$

i. Decorrelating Rake Receiver (DRR) Algorithm

Received signal at base band is given by

$$y(t) = \sum_{k=1}^{n} A_k b_k S_k(t) + \sigma n(t) \ldots\ldots \quad (5)$$

Where,

$S_k(t)$ = deterministic signature waveform assigned to the k^{th} user, normalized so as to have unit energy

A_k = received gain of the linear time invariant channel for user k.

b_k = bit transmitted by the k^{th} user values must be either1's and -1's

n(t) = white Gaussian noise with unit power spectral density. The sampled output of the matched filter for k^{th} user is

$$Y_k = \int_0^T Y_t . S_k(t) . dt \qquad (6)$$

The output of the each matched filter can be represent as $y_1[t], y_2[t], \ldots, y_k[t]$.Here $y_1[t]$ is the output of the first matched filter and so on.

In Matrix representation, output of the Matched filter is given by

$$Y = R.A.b + n \qquad (7)$$

So the decision of the k^{th} user is made based on

$$\hat{b}_k = \mathrm{sgn}((R^{-1}y)_k)$$

$$\hat{b}_k = \mathrm{sgn}(R^{-1}(RAb + n)_k)$$

$$\hat{b}_k = \mathrm{sgn}((Ab + R^{-1}n)_k) \qquad (8)$$

The output of the Decorrelator Detector is

$$R^{-1}y = R^{-1}RAb + R^{-1}n$$

$$R^{-1}y = Ab + R^{-1}n \qquad (9)$$

ii. Algorithm for MMSE Detector

a) The received signals are demodulated using BPSK Demodulation technique.

b) These symbols are then despreaded to be decoded.

c) The following filters like Matched filter, Decorrelating Matched filter and MMSE [4] [7] are applied and corresponding bits are decoded.

d) The received bits are then compared with the original symbols and the corresponding BER is computed.

So the decision for the k^{th} user is made based on

$$\hat{b}_k = \mathrm{sgn}(((R + \sigma^2 A^{-2})^{-1} y)_k)$$

$$\hat{b}_k = \mathrm{sgn}\left(\left((R + \sigma^2 A^{-2})^{-1}(RAb + n)\right)_k\right)$$

III. ADAPTIVE DECISION EQUALIZER

Adaptive equalization [10] is the technique used to reliably transmit data through a communication channel. Ideally, if the channel is ideal (without and channel distortion and additive noise), we can demodulate the signal perfectly at the output without causing any error. However, in practice, all the channels are non-ideal and noisy in nature. So, to recover the original signal after demodulation,

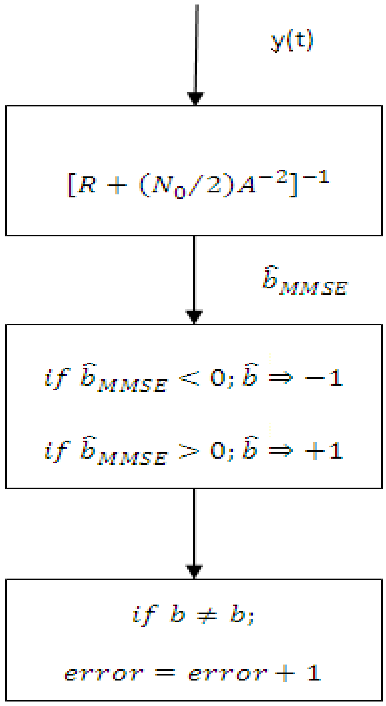

Fig 2. Algorithm for MMSE Detector

our aim is to find an equalization filter which will minimize the error between original transmitted signal and demodulated signal passed through equalization filter. Several algorithms like Least Mean Square (LMS).In this project, we study the adaptive equalization technique with the use of least mean Square algorithm.

Channel Equalization

The inter symbol interference imposes the main obstacles to achieving increased digital transmission rates with the required accuracy. ISI problem is resolved by channel equalization [9] in which the aim is to construct an equalizer such that the impulse response of the channel/equalizer combination is as close to $Z^{-\Delta}$ as possible, where Δ is a delay. Frequently the channel parameters are not known in advance and moreover they may vary with time, in some applications significantly. Hence, it is necessary to use the adaptive equalizers, which provide the means of tracking the channel characteristics.

In Fig 3 s(n) is the signal that you transmit through the communication channel, and x(n) is the distorted output signal. To compensate for the signal distortion, the adaptive channel equalization system as shown in Fig 3 completes the following two modes:

a) Training mode

This mode helps you determine the appropriate coefficients of the adaptive filter. When you transmit the signal s(n) to the communication channel, you also apply a delayed version of the same signal to the adaptive filter. In the previous Fig, $Z^{-\Delta}$ is a delay function and d(n) is the delayed signal, y(n) is the output signal from the adaptive filter and e(n) is the error signal between d(n) and y(n). The adaptive filter as shown in

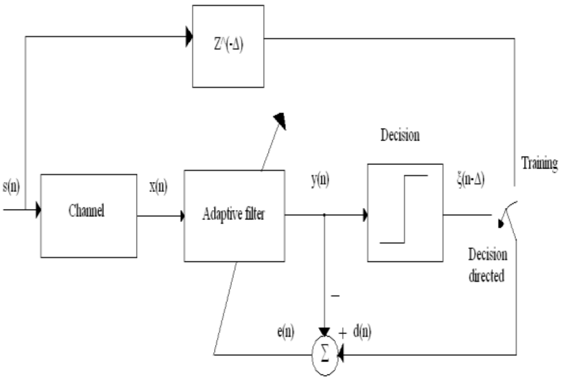

Fig 3. Digital transmission system using channel equalization

Fig 3.5 iteratively adjusts the coefficients to minimize e(n) .After the power of e(n) converges, y(n) is almost identical to d(n),which means that you can use the resulting adaptive filter coefficients to compensate for the signal distortion.

b) Decision-directed mode

After you determine the appropriate coefficients of the adaptive filter, you can switch the adaptive channel equalization system to decision-directed mode. In this mode, the adaptive channel equalization system shown in Fig 4 decodes the signal and y(n) produces a new signal, which is an estimation of the signal s(n) except for a delay of Δ taps.

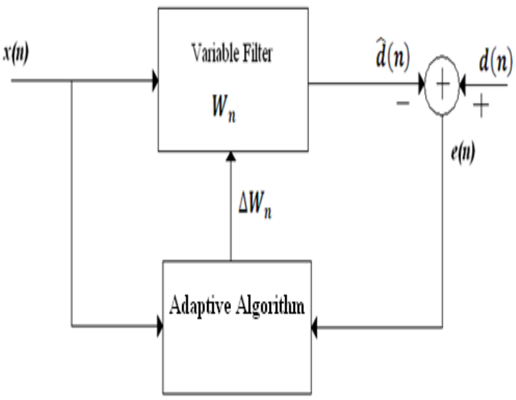

Fig 4. Adaptive filter

The input signal is the sum of a desired signal *d (n)* and interfering noise *v (n)*.

$$x(n) = d(n) + v(n) \quad (10)$$

The variable filter has a Finite Impulse Response (FIR) structure. For such structures the impulse response is equal to the filter coefficients. The coefficients for a filter of order p are defined as

$$w_n = [w_n(0), w_n(1), \ldots, w_n(p)]^T \quad (11)$$

The error signal or cost function is the difference between the desired and the estimated signal

$$e(n) = d(n) - \hat{d}(n) \quad (12)$$

The variable filter estimates the desired signal by convolving the input signal with the impulse response. In vector notation this is expressed

$$\hat{d}(n) = w_n * x(n) \quad (13)$$

Where,

x(n)=〚[x(n),x(n-1),….,x(n-p)]〛^T

is an input signal vector. Moreover, the variable filter updates the filter coefficients at every time instant

$$W_{n+1} = W_n + \Delta W_n \quad (14)$$

Where, ΔW_n is a correction factor for the filter coefficients .The adaptive algorithm generates this correction factor based on the input and error signals.

Two adaptive methods which employ this least square error penalty function are the least mean square (LMS) and the more complex recursive least squares (RLS) algorithms. LMS algorithm [13] is depicted schematically in Fig 5.

In LMS algorithm, correlation with an FIR filter is performed to obtain a (soft) estimate, $\hat{x}$, of the training data bit x(n) , as in the correlator receiver. The error e(n) in this estimate is then used to update the tap weights of the FIR receiver filter. In the LMS algorithm, this is performed by simple weighting of the error by step size μ.

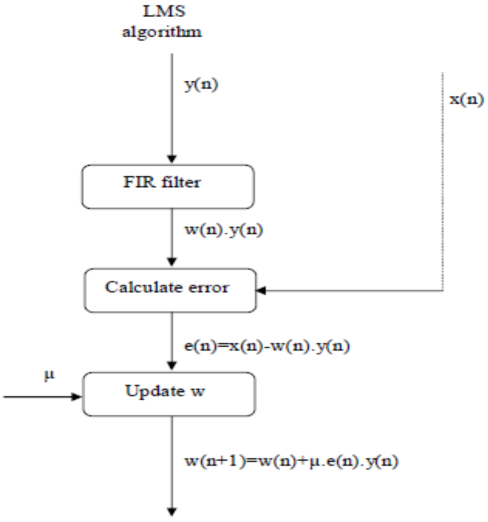

Fig 5. LMS algorithm

IV. RESULTS

The simulated results are presented here for rake receiver, de-correlating rake and MMSE detector and Adaptive decision equalizer. In the simulation the spreading factor of {16 32 64} for different user considered. The channel weights were considered to be of minimum phase whose coefficient are {1 0.5 0.2}. At a time total 80 symbols were transmitted in each slot. Here we have used the BER as a parameter for the performance evaluation. Fig 6 presents the result of BER vs. SNR for two users. At 40dB we get an improved performance of MMSE detector compared to rake receiver. In Fig.7 shows the performance for two users with a spreading factor of 32 and here we get a MMSE detector performance better than rake receiver which means even if the number of user increases then also we get a better result in MMSE detector compared to decorrelating rake. Similarly Fig 9, Fig 10 and Fig 11 show the BER performance for five users respectively for a spreading factor of 16, 32, and 64. In the Fig we find that as SNR increases rake receiver degrades while MMSE detector gives a better performance. ADE has better performance when compare to rake receivers proposed. BER for ADE is low for lesser SNR value because it uses adaptive techniques to improve signal strength at receiver.

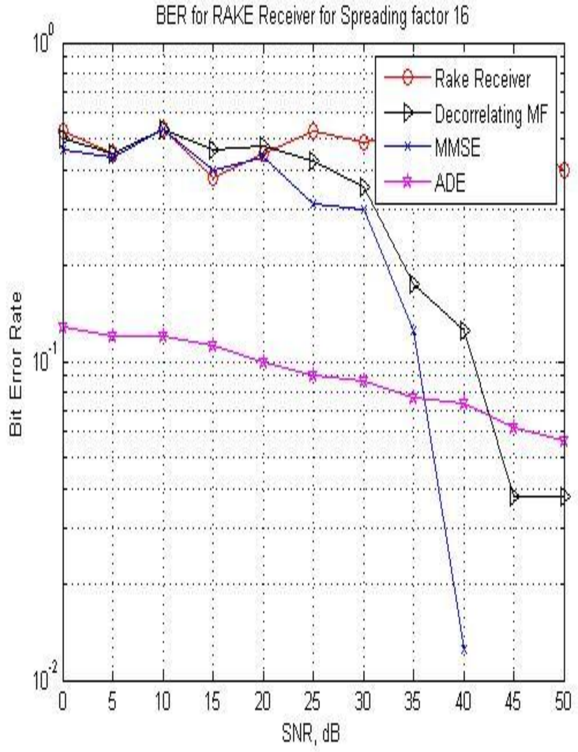

Fig 6. BER vs SNR for 2 users at a Spreading factor of 16

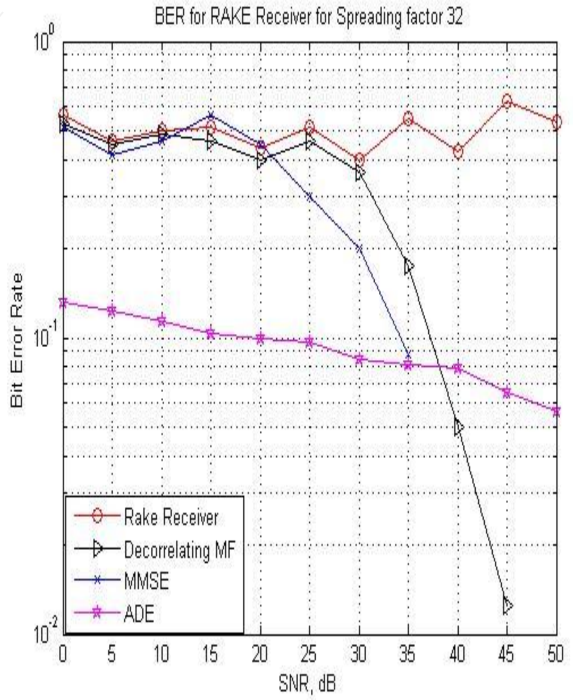

Fig 7. BER vs SNR for 2 users at a Spreading factor of 32

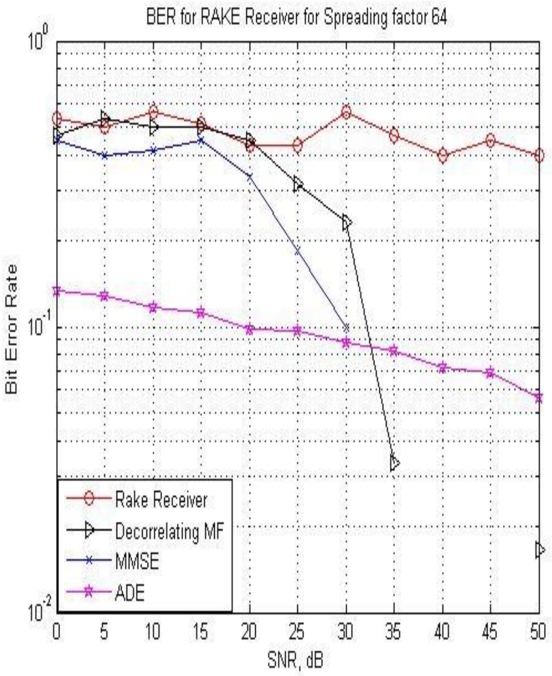

Fig 8. BER vs SNR for 2 users at a Spreading factor of 64

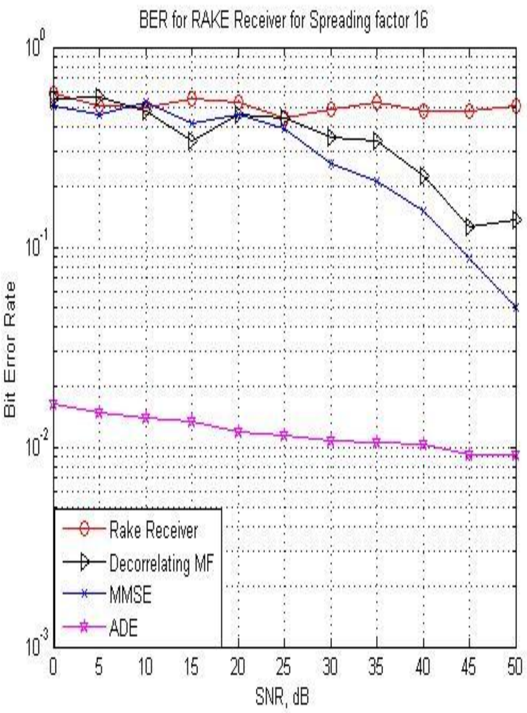

Fig 9. BER vs SNR for 5 users at a Spreading factor of 16

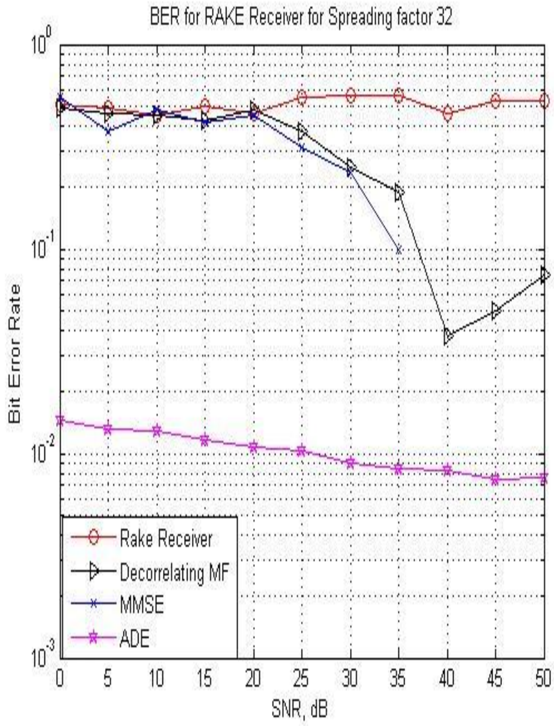

Fig 10. BER vs SVR for 5 users at a Spreading factor of 32

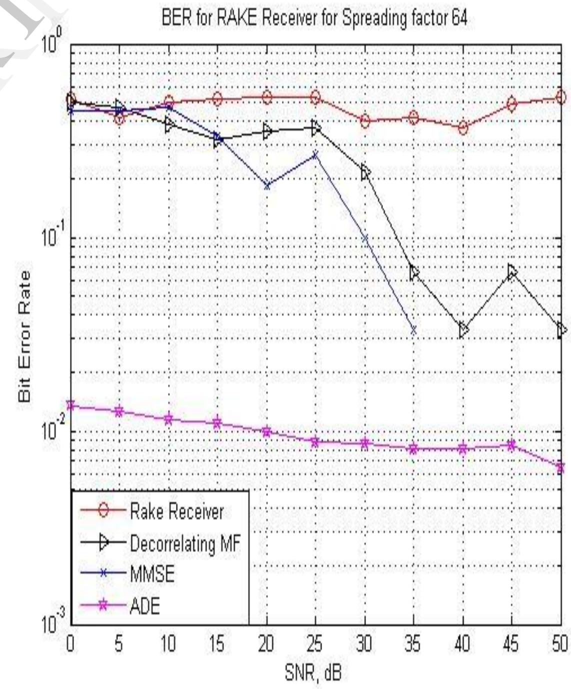

Fig 11. BER vs SNR for 5 users at a Spreading factor of 64

V. CONCLUSION

The DS-CDMA Rake Receiver is realized using BPSK Modulation and Demodulation techniques. Matched filter, De-correlating receiver and Minimum mean square error equalizer (MMSE) are realized at the receiver side and their corresponding BER are plotted against different SNR values. It has been observed that for different no of users, same spreading factor, and the BER decreases at a faster rate for the MMSE detector. We find that as SNR increases rake receiver degrades, de-correlating rake receiver while MMSE detector gives a better performance. An adaptive decision equalizer using LMS algorithm response also plotted. From the above results, as no of users increases the bit error rate increases. When compare to conventional Rake receiver and decorrelating MF, detector using equalizer has better performance.

VI. FUTURE SCOPE

By using Linear MMSE detector the ISI can be reduce, but it is considered that channel is AWGN channel. In AGWN, noise is uniform and white noise. We can implement for color noise where noise is not uniform. We can apply non linear adaptive techniques to received signals so that we can achieve better performance.

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