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Adaptive Estimation of the Doppler Shift in CDMA Using Monte Carlo Method

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ABSTRACT

This paper presents the startagy of applying the concept of CDMA to the MIMO OFDM System for the security of the information and also to calculate the bit error rate of the system in the Rayleigh fading and rician frequency selective fading channels.The system actually uses space time block coding's for the different configurations of transmit and receive antennas. The channel estimation is performed by pilot sequences at each transmit antenna. The simulation results are compared with MIMO OFDM, OFDM CDMA downlink system . The proposed system shows that the presented system provides low BER, this system is applicable to 4g Wireless networks.

Keywords: CDMA, DS-CDMA, EM, FDE, SMC-KF

I. INTRODUCTION

An Adaptive frequency estimation algorithm to estimate the Doppler shift in DS-CDMA with multiple Doppler sub paths. Rake receivers have been widely used in Direct Sequence Code Division Multiple Access (DS-CDMA) radios to achieve multipath diversity. However, a traditional rake receiver has a high computational complexity and offers limited capability to exploit the inherent diversity nature of doubly selective channels. In particular, it suffers serious performance degradation in fast fading channels. To enhance performance of a DS-CDMA radio for wireless applications, fast fading should be mitigated by exploring the inherent diversity nature of the doubly selective channels .To achieves this goal, a concept of Doppler diversity was proposed. The theoretical basis of Doppler diversity lies on the pseudo-random characteristics of pseudo noise (PN) codes used in DS-CDMA radios that ensure the orthogonality of different Doppler sub paths.

To improve the performance of a DS-CDMA radio receiver in a cost-effective manner, frequency domain equalization (FDE) has been considered in many previous works. However, those studies often involved complicated structures, such as the multiple-tap time-domain feedback filter, etc. It is desirable to develop an efficient channel estimation algorithm that can work closely with the FDE to utilize Doppler diversity in a doubly selective channel to improve receiver's performance.

Based on the Doppler shift estimation, an innovative FDE based receiver architecture is proposed to take advantages of Doppler diversity in fast fading channels of a DS-CDMA radio. Moreover, this architecture can easily incorporate a frequency domain channel estimation algorithm to avoid using a costly rake receiver.

Iterative algorithms, such as Expectation-Maximization (EM) algorithms, have been widely used in Doppler shift estimation. Along with that the BER performance is also analyzed under various SNR conditions of FDE receiver with Doppler shifts. Monte Carlo analysis is done to improve better error performance; it is done by using Sequential Monte Carlo- Kalman Filter algorithm (SMC-KF). Simulation results are included.

DOOPLER SHIFT

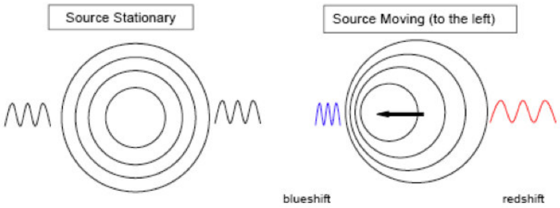


Figure 1.1 Change of wavelength caused by motion of the source

The Doppler effect (or Doppler shift), named after the Austrian physicist Christian Doppler, who proposed it in 1842 in Prague, is the change in frequency of a wave moving relative to its source. It is commonly heard when a vehicle sounding a siren or horn approaches, passes, and recedes from an observer. The received frequency is higher (compared to the emitted frequency) during the approach, it is identical at the instant of passing by, and it is lower during the recession

The relative changes in frequency can be explained as follows. When the source of the waves is moving toward the observer, each successive wave crest is emitted from a position closer to the observer than the previous wave. Therefore each wave takes slightly less time to reach the observer than the previous wave. Therefore the time between the arrivals of successive wave crests at the observer is reduced, causing an increase in the frequency. The distance between successive wave fronts is increased, so the waves "spread out". For waves which do not require a medium, such as light or gravity in general relativity, only the relative difference in velocity between the observer and the source needs to be considered

2. Frequency Domain Equalization Receiver

Based on estimated Doppler shift result, a frequency domain equalizer (FDE) based receiver architecture is developed to exploit Doppler diversity in frequency domain. This receiver architecture is used to achieve low complexity and good performance compared to traditional CDMA. To improve performance of DS-CDMA in cost effective manner FDE is used. channel coding and frequency-domain interleaving which are necessary for OFDM systems on multipath fading channels characterized by deep notches in the signal spectrum.

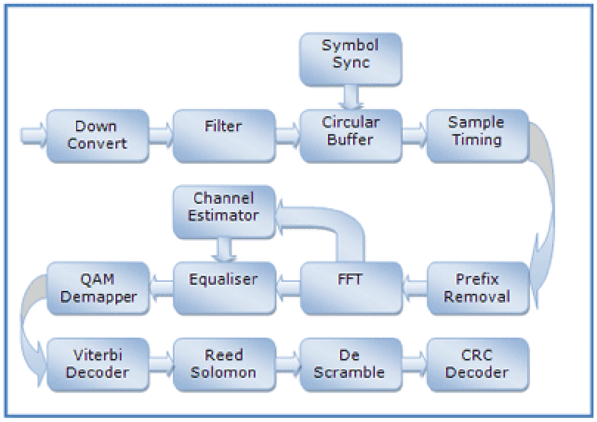


FIGURE 2.1 FDE RECEIVER IN MULTI-CARRIER SYSTEM

2.1 EM ALGORITHM

The EM algorithm is used to find the maximum likelihood parameters of a statistical model in cases where the equations cannot be solved directly.The EM iteration alternates between performing an expectation (E) step and maximization (M) step, E step which creates a function for the expectation of the log-likelihood evaluated using the current estimate for the parameters M step, which computes parameters maximizing the expected log-likelihood found on the *E* step. These parameter-estimates are then used to determine the distribution of the latent variables in the next E step. Latent variable is basically physical quantity variable ex: channel, tx model. It is hidden variable.

2.2 KALMAN FILTER

Filtering is desirable in many situations in engineering. For example, radio communication signals are corrupted with noise. A good filtering algorithm can remove the noise from electromagnetic signals while retaining the useful information. The Kalman filter is a tool that can estimate the variables of a wide range of processes. In mathematical terms we would say that a Kalman filter estimates the states of a linear system. The Kalman filter not only works well in practice, but it is theoretically attractive because it can be shown that of all possible filters, it is the one that minimizes the variance of the estimation error.

2.3 MONTE CARLO METHOD

Monte Carlo methods are a widely used class of computational algorithms for simulating the behavior of various physical and mathematical systems, and for other computations. They are distinguished from other simulation methods (such as molecular dynamics) by being stochastic, that is nondeterministic in some manner – usually by using random numbers (in practice, pseudo-random numbers) – as opposed to deterministic algorithms. Because of the repetition of algorithms and the large number of calculations involved, Monte Carlo is a method suited to calculation using a computer, utilizing many techniques of computer simulation.

An analytical technique in which a large number of simulations are run using random quantities for uncertain variables and looking at the distribution of results to infer which values are most likely

A Monte Carlo algorithm is often a numerical Monte Carlo method used to find solutions to mathematical problems (which may have many variables) that cannot easily be solved, for example, by integral calculus, or other numerical methods. For many types of problems, its efficiency relative to other numerical methods increases as the dimension of the problem increases. Or it may be a method for solving other mathematical problems that rely on pseudo-random numbers.

2.4 APPLICATION OF MONTE CARLO

Monte Carlo simulation methods are especially useful in studying systems with a large number of coupled degrees of freedom, such as liquids, disordered materials, strongly coupled solids, and cellular structures. More broadly, Monte Carlo methods are useful for modeling phenomena with significant uncertainty in inputs, such as the calculation of risk in business (for its use in the insurance industry, see stochastic modeling). A classic use is for the evaluation of definite integrals, particularly multidimensional integrals with complicated boundary conditions.

Monte Carlo methods are very important in computational physics, physical chemistry, engineering, corporate finance, and related applied fields, and have diverse applications from complicated quantum chromo dynamics calculations to designing heat shields and aerodynamic forms.

3.ADAPTIVE DOPPLER SHIFT ESTIMATION

Adaptive Estimation algorithm to estimate the Doppler shift for each Doppler sub path. The method works based on the expectation-maximization (EM) algorithm. This method forms a basis to implement a frequency domain equalizer, such that the Doppler diversity can be achieved. In the following, we consider a scenario with L = 4, Q = 1. However, it is easy to extend the discussions to any other scenarios.

Tapped delay line channel model is expressed as

$$h[n;T]=\sum_{l=1}^{L} h_l\,(n)\,\delta\left(T-\left[\frac{Tl}{Tc}\right]\right) \quad (1)$$

Where $[\frac{Tl}{Tc}]$ is the largest integer smaller than $\frac{Tl}{Tc}$,

Tl denotes the delay of the l^{th} sub path

Tc is the chip duration

h*l (n)* is the discrete complex channel coefficient of the *l*th sub path

L stands for the number of sub path

In the aforementioned channel model, the received pilot signal in a typical DS-CDMA radio can be expressed as

$$r(n)=\sum_{l=1}^{L} h_l\,(n)a\left(T-\left[\frac{Tl}{Tc}\right]\right)+N_r(n) \qquad (2)$$

Nr (n) denotes the additive white Gaussian noise (AWGN) at the receiver. Without loss of generality, we assume $|a(n)|$ =1.

3.1 BLOCK DIAGRAM

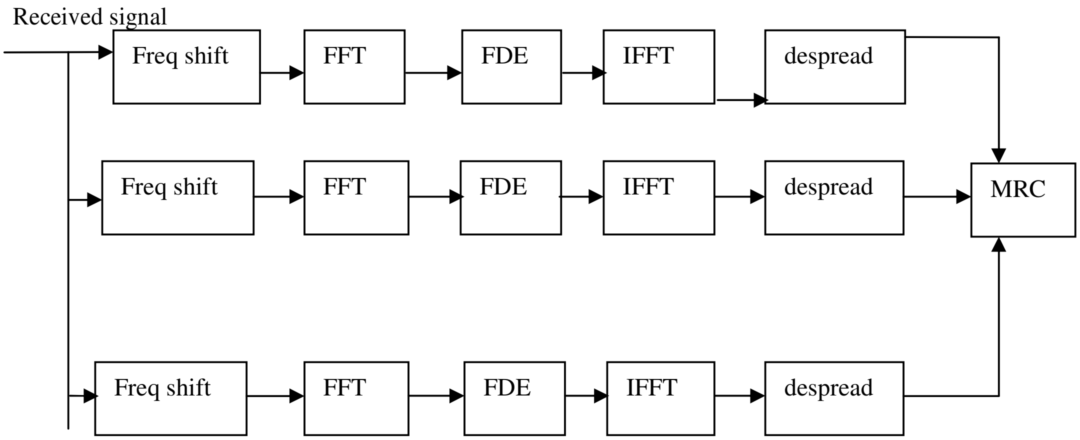


FIGURE 3.1 PROPOSED FDE-BASED RECEIVER

3.2 ESTIMATION OF TIME INVARIANT COEFFICIENTS

The number of significant sub paths is four (L = 4). The time-invariant coefficient matrix is given by

$$\bar{H}_{BEM}=\begin{bmatrix} h_{-1,1} & h_{0,1} & h_{1,1} \\ h_{-1,2} & h_{0,2} & h_{1,2} \\ h_{-1,3} & h_{0,3} & h_{1,3} \\ h_{-1,4} & h_{0,4} & h_{1,4} \end{bmatrix} \qquad (3)$$

n denotes the index of the discrete time sampled at a period of *Tc* and L is the total number of channel taps. Totally there is an L X (2Q+1) sub paths, Nmod should be at least larger than the length of the transmitted data frame. we choose Q=1,Nmod can be empirically set to be 3N

Received signal, based on the prior knowledge of the pilot signal, a correlation operation is performed at the receiver side as,

$$r(n)=\sum_{l=1}^{L}\sum_{q=-Q}^{Q} h_{q,l}e^{j2\pi qn/N_{mod}}a(n-Tc)+N_r(n) \qquad (4)$$

$$y\,(m)=\frac{1}{W}\sum_{n=1}^{W} r(n+m)a^*(n)$$

$$=\sum_{n=1}^{W}\sum_{l=1}^{L}\sum_{q=-Q}^{Q}\frac{h_{ql}}{W}e^{j2\pi q(n+m)/Nmod}a(n+m-Tl)a^*(n)+z\,(m)$$

Where m = 0, 1... M, W is chosen autocorrelation window size and M is the upper bound of the search range of the autocorrelation operation. Z (m) denotes additive noise at the output of the correlate. Equation (5) may be used to calculate the correlation (correlation range up to W) between pilot symbol a (n) and the received signal r (n+ m), which is m time shifted version of r (n) (shift range up to M).

To extract channel information, the time-shifted version of the received signal r (n) is correlated with the known pilot signal a (n) in a DS-CDMA radio. Let us consider the first sub path where *l* =1 and assume m = T1, Q=1 and $|a(n)|$=1.based on the pseudo-orthogonality features of a (n), equation (4) can be reduced to

$$Y(T1,W)\approx\sum_{n-1}^{W}\sum_{q--1}^{1}\frac{h_{q,l}}{W}e^{j2\pi q(n+T1)/Nmod}+z(Tl) \qquad (6)$$

3.3 ESTIMATION OF DOPPLER SHIFTS

In this subsection, an adaptive estimation scheme is pro- posed. With both time-varying and time-invariant information of a doubly selective channel, a complete description of a fast fading channel can be established. Utilizing the estimated time-invariant coefficient matrix given in equation (3), we can obtain the frequency response of the doubly selective channel as

$$\bar{H}=\mathrm{diag}\{\overline{TH}\,\mathrm{BEM}\bar{\gamma}\} \qquad (7)$$

Based on the BEM model, the signal through the channel r (n) is given by. Although noise may influence the performance of an estimator, the additive noise in does not affect the derivation of the Doppler shift estimation algorithm and can thereby be neglected in the following analysis. The Fourier Transform of r (n), without considering the additive noise term, can be expressed as

$$R\,(k)=\sum_{n=0}^{p\,-1} r(n)e^{-j2\pi nk/p}$$

$$=\sum_{n=0}^{p-1}\sum_{l=1}^{L}\sum_{q=-Q}^{Q} hq,la(n-T1)e^{\frac{-j2\pi n}{p}}\left(k-\frac{qP}{Nmod}\right) \qquad (8)$$

Therefore, an improved estimation on the Doppler frequency shift is necessary. Let fe denote the unknown estimation error of Doppler shift can be modifed as

$$R(k) = \sum_{n=0}^{p-1} r(n) e^{-j2\pi nk/p}$$

$$= \sum_{n=0}^{p-1} \sum_{l=1}^{L} \sum_{q=-Q}^{Q} hq,la(n - T1)e^{\frac{-j2\pi n}{P}}\left(k - \frac{qP}{Nmod}\right) \quad (9)$$

Where qp/Nmod can be viewea as initial estimated Doppler frequency shift from the BEM channel model,which is predetermined by Nmod.that is ,when Q=1,the initial Doppler shift estimation for the Doppler sub paths are simply

{-p/Nmod, 0, p/Nmod}

Let fe denote the unknown estimation error of Doppler shift .equation (10) can be modified as

$$R(k) = \sum_{n=0}^{p-1} r(n) e^{-j2\pi nk/p} \quad (10)$$

$$= \sum_{n=0}^{p-1} \sum_{l=1}^{L} \sum_{q=-Q}^{Q} hq,la(n - T1)e^{\frac{-j2\pi n}{P}}\left(k - \frac{qP}{Nmod} + fe\right)$$

That is, R (k) is the Fourier Transform f the recovered signal with Doppler shift estimation error. On the other hand, the Fourier Transform of the received signal $\tilde{r}$(n) (the signal through a physical channel) can be approximated by

$$\tilde{R}(k) = \sum_{n=0}^{p-1} \tilde{r}(n) e^{-j2\pi nk/p}$$

$$\sum_{n=0}^{p-1} \sum_{l=1}^{L} \sum_{q=-Q}^{Q} \tilde{h}_{q'} la(n - T1)e^{\frac{-j2\pi n}{P}}(k + fe(q)) \quad (11)$$

The real part of the correlation of R (k) and $\tilde{R}$(k) can be approximated by a quadratic form as

$$R\{E[R^*(k)\tilde{R}(k)]\} \approx A(fe)^2 + Bfe + C \quad (12)$$

Where constants A, B, and C are given by

$$A = -R\left\{\sum_{l=1}^{L}\sum_{q1=-1}^{1}\sum_{q2=-1}^{1}\ddot{h}_{q1,l}\, h *_{q1,l}\right\}\sum_{n=n}^{p-1}\frac{2\pi^2 n^2}{p^2}$$

$$B = R\left\{\sum_{l=1}^{L}\sum_{q=-1}^{1}\ddot{h}_{-1,l}\, h *_{q,l}\right\}\sum_{n=0}^{p-1}\frac{4\pi^2 n^2}{p^2} fe(-1)$$

$$+ R\left\{\sum_{l=1}^{L}\sum_{q=-1}^{1}\ddot{h}_{0,l}\, h *_{q,l}\right\}\sum_{n=0}^{p-1}\frac{4\pi^2 n^2}{p^2} fe(0)$$

$$+ R\left\{\sum_{l=1}^{L}\sum_{q=-1}^{1}\ddot{h}_{1,l}\, h *_{q,l}\right\}\sum_{n=0}^{p-1}\frac{4\pi^2 n^2}{p^2} fe(1)$$

$$+ R\left\{\sum_{l=1}^{L}\sum_{q1=-1}^{1}\sum_{q2=-1}^{1}\ddot{h}_{q2,l}\, h *_{q1,l\frac{qP}{Nmod}}\right\}\sum_{n=0}^{p-1}\frac{4\pi^2 n^2}{p^2}$$

$$- R\left\{\sum_{l=1}^{L}\sum_{q1=-1}^{1}\sum_{q2=-1}^{1}\ddot{h}_{q2,l}\, h *_{q1,l}\right\}\sum_{n=0}^{p-1}\frac{2\pi n}{p}$$

$$= B_1 fe(-1) + B_2 fe(0) + B_3 fe(1) + B_4 \quad (13)$$

The expression for constant C is not given because it will not be involved in the remaining process. It is noted that in calculating the estimator coefficients A and B,$\ddot{h}_{q,l}$ and $h_{q,l}$ are assumed to have the same value, which can be found in the matrix $\overline{H}_{BEM}$ in equation(3).with the original received signal, we can get the value of fe , noted as fe1,via an exhaustive search, getting equation (12) reach its maximum value. In addition, as A will have a non-zero value, it is well known that when *f*e = $-\frac{B}{2A}$, equation (12) will have the maximum value. Therefore will have

$$-\frac{B}{2A} = -\frac{B_1}{2A} fe(-1) - \frac{B_2}{2A} fe(0) - \frac{B_3}{2A} fe(1) - \frac{B_4}{2A} = f_{e1} \quad (14)$$

Using similar approach, through circularly shifting the columns of $\overline{H}BEM$ to the left,we can obtain different estimations of the received signal r (n),which can be used to construct the other two linear equations, or

$$-\frac{B_2}{2A} fe(-1) - \frac{B_3}{2A} fe(0) - \frac{B_1}{2A} fe(1) - \frac{B_4}{2A} = f_{e2} \quad (15)$$

$$-\frac{B_3}{2A} fe(-1) - \frac{B_2}{2A} fe(0) - \frac{B_1}{2A} fe(1) - \frac{B_4}{2A} = f_{e3} \quad (16)$$

3.4 ALGORITHM GENERALIZATION

The estimation algorithm can be extended to include more Doppler sub paths and time domain sub paths. In general, if the number of Doppler sub paths of interest is denoted by 2Q + 1 and the number of time domain sub paths is L, the number of autocorrelation window sizes becomes Q ([W1;W2; …. ;WQ]), and the size of the time-invariant channel coefficient matrix ($\overline{H}$BEM) increases to LxQ. Moreover, the lengths of equations increase to L x Q x P. Except for the increased complexity, the algorithm procedure remains to be the same.

Assume that the estimated synchronization parameter is n_d after the initial estimation. The transmitted signal over a particular Doppler sub path can be equivalently expressed by $(n - n_d)$. The P-point DFT of the transmitted signal is given by

$$S(K) = \sum_{n=0}^{P-1} a(n - nd) e^{\frac{-j2\pi kn}{P}} \quad (17)$$

With the assumption of $m = n - n_d$, the above equation can be written as

$$S(K) = \sum_{m=-n_d}^{P-1-n_d} a(m) e^{\frac{-j2\pi k(m+n_d)}{P}} \quad (18)$$

Then DFT sequence of the equivalent transmitted signal affected by the synchronization error can be expressed b

$$\bar{S} = \bar{\Omega}_{nd}\,\bar{A} \quad (19)$$

Under the assumption of $n_e = n_d - n_c$, where n_c is the coarse estimation of synchronization parameter and n_e denotes the estimation error in each iteration, equation (19) can be rewritten as

$$\bar{S} = \bar{\Omega}_{ne}\,\bar{F} \quad (20)$$

3.5 ADAPTIVE ESTIMATION ALGORITHM:

Step1: Set iteration index i=1

Step2: Obtain estimated received signal R (K)

Step3: Obtain cross correlation of R (K) and $\tilde{R}(K)$

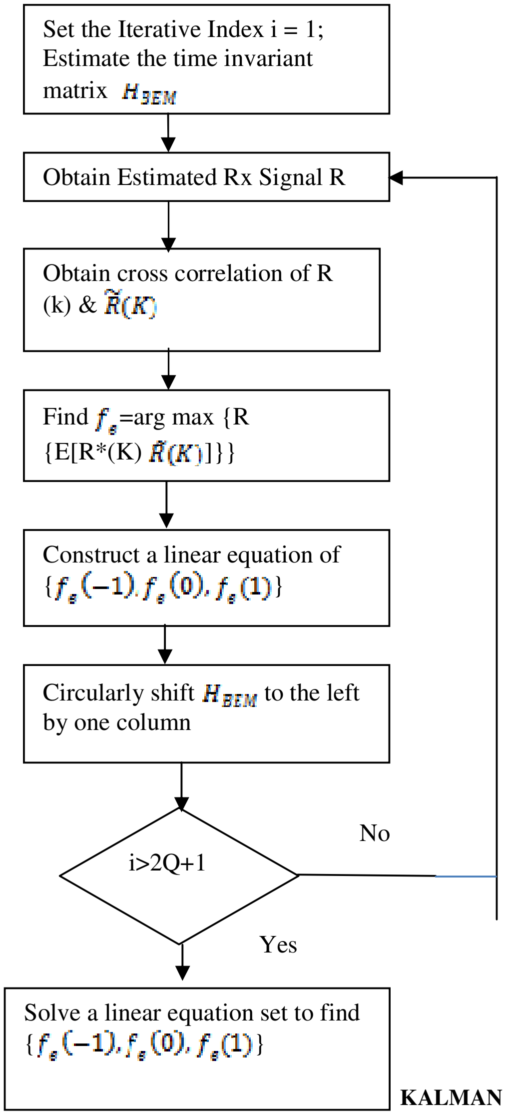
Step4: Find fe

Step5 : Construct linear equation{fe(-1),fe(0),fe(1)}

Step6: Solve linear equation

A sequential Monte Carlo filter is considered which combines previously developed Sequential Importance Sampling (SIS) techniques for conditional linear Gaussian models with measurement linearization for construction of approximate simulation densities The resulting sequential Monte Carlo Kalman Filter (SMC-KF) consists of a bank of conventional Kalman filters individually tuned to sampled trajectories of the nonlinear state variables.

3.6 FLOWCHART



KALMAN FILTER ALGORITHM

For i=1, 2….P

Given: $P_i(n|n-1), \hat{X}_i(n|n-1), C_i(n-1)$

Step1: Compute Jacobian matrix

$$J_i(n) = \frac{\partial H(c)\hat{X}_i(n|n-1)}{\partial c} \qquad c = F_c c_i(n-1)$$

Step2: Compute approximate innovations covariance matrix

$$\Sigma_i(n|n-1) =$$

$$H\,(F_{cC_i}(n-1)P_i(n|n-1)H(F_{cC_i}(n-1))^H + R$$

Step3: Compute sampling distribution mean/covariance

$$P_{c,i}(n)^{-1} = Q_c^{-1} + J_i(n)^H \Sigma_i(n|n-1)^{-1} J_{i(n)}$$

$\bar{C}_i(n) = F_c C_i(n-1) + P_{c,i}(n) J_i(n)^H \Sigma_i(n|n-1)^{-1}$[r(n)-H($F_c C_i(n-1))\hat{X}_i(n|n-1)$]

Step4: Sample $C_i(n) \sim$ N $(C_i(n); \bar{C}_i(n), P_{c,i}(n))$

Step5: Compute Kalman updates

$$P_i(n|n)^{-1} = P_i(n|n-1)^{-1} + H(C_i(n))^H R^{-1} H(C_i(n))$$

$$\hat{X}_i(n|n) = \hat{X}_i(n|n-1) + P_i(n|n)H(C_i(n))^H R^{-1}[r - H(C_i(n))\hat{X}_i(n|n-1)]$$

$$\hat{X}_i(n+1|n) = F_x \hat{X}_i(n|n)$$

$$P_i(n+1|n) = F_x P_i(n|n) F_x^H + Q_x$$

Step6: Next i

4. SIMULATION RESULTS

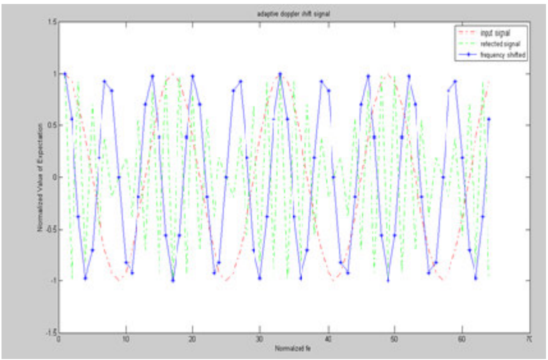


FIGURE 4.1 DOPPLER SHIFT

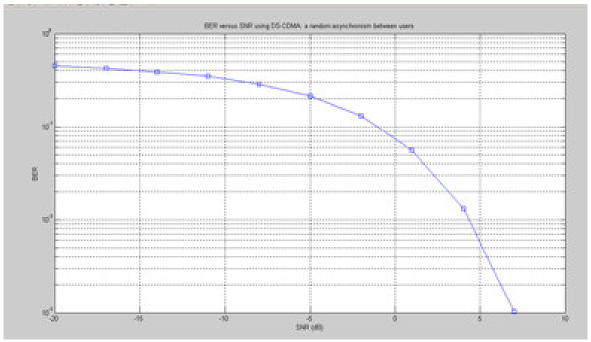


FIGURE 4.2 BER PERFORMANCE USING HADAMARD MATRIX

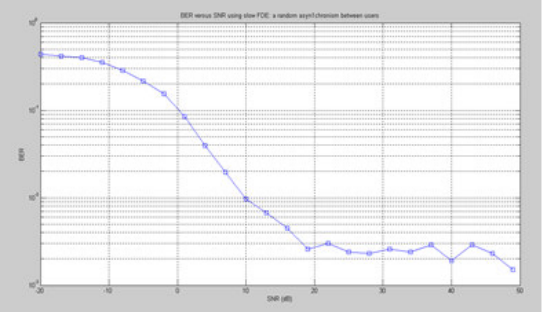


FIGURE 4.3 BER PERFORMANCE WITHOUT HADAMARD MATRIX

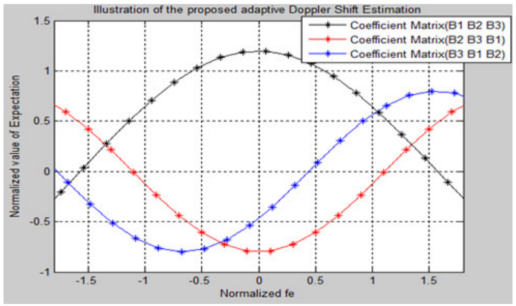


FIGURE 4.4 ADAPTIVE DOPPLER SHIFT ESTIMATION

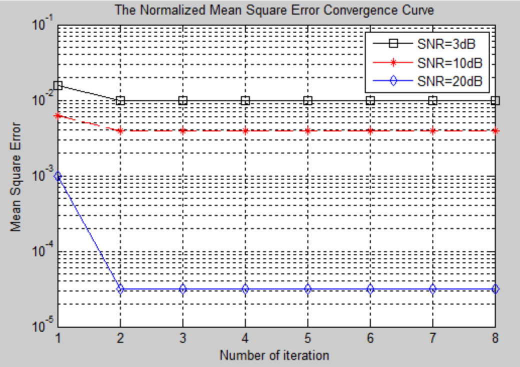


FIGURE 4.5 NORMALIZED MEAN SQUARE ERRORS

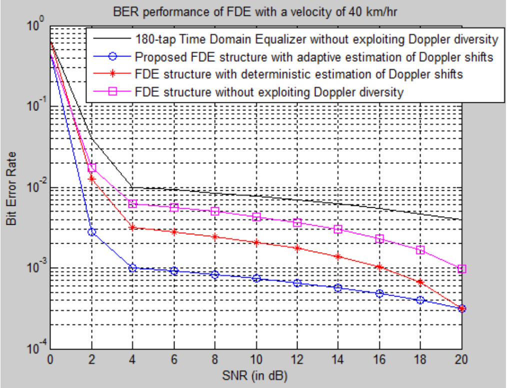


FIGURE 4.6 BER PERFORMANCE OF FDE RECEIVER OF VELOCITY 40KM/HR

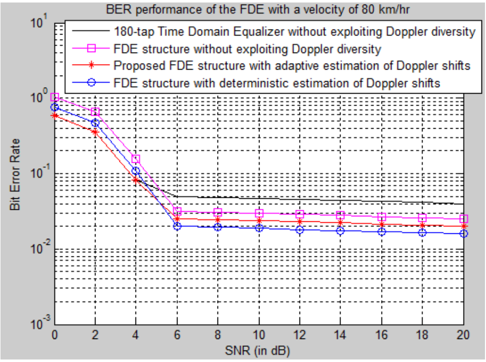


FIGURE 4.7 BER PERFORMANCE OF FDE RECEIVER OF VELOCITY 80KM/HR

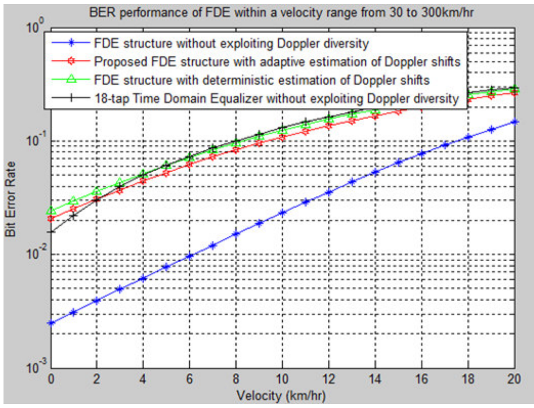


FIGURE 4.8 PERFORMANCE OF BER VS. VELOCITY RANGE FROM 30 TO 300KM/HR

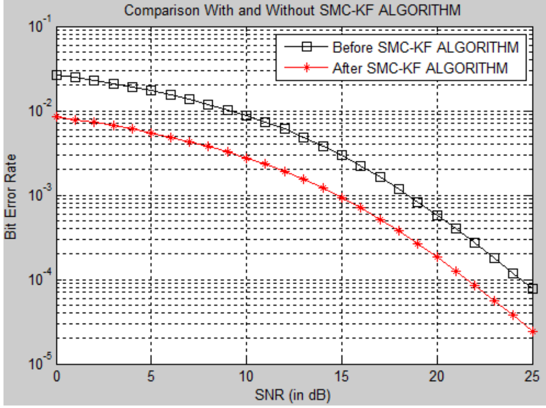


FIGURE 4.9 COMPARISON OF WITH AND WITHOUT SMC-KF ALGORITHM

5 .CONCLUSION

In this project, adaptive frequency estimation algorithm is proposed. The aim purposes of the presented algorithms are to estimate Doppler shift in Direct Sequence Code Division Multiple Access (DS-CDMA) with Doppler sub path. The main idea was to estimate Doppler shifts. By using an Expectation-Maximization (EM) algorithm based adaptive estimation method is developed to extract accurate Doppler shift information. EM iteration alternates between Expectation (E) and Maximization (M) step. E step creates function for the expectation of log-likelihood evaluated using current estimate for the parameters. M step, which compute parameter maximizing the expected log-likelihood found on E-step.

Based on the estimated Doppler shift results, Frequency Domain Equalizer (FDE) based receiver architecture is developed to exploit Doppler diversity in the frequency domain. Error performance of Bit Error Rate (BER) vs. Signal to Noise Ratio (SNR) of DS-CDMA with FDE structure Doppler shift was estimated. The advantage of our proposed method gives better performance, less complexity and least BER .By comparing error performance of DS-CDMA with FDE structure gives better BER. Here we are considering only antenna and channel coding gain but while we using Doppler shift, channel coding and antenna gain will lose its performance so at finally to improve system with properly selective channel coding, Monte-Carlo method is used order to improve properly selective channel by using Sequential Monte Carlo-Kalman Filter (SMC-KF) algorithm. SMC-KF algorithm gives better error performance comparing to adaptive estimation algorithm.

6 .REFERENCES

[1] O.Rousseaux,G.Leus, and M.Moonen, ”Estimation And Equalization Of Doubly Selective Channels Using Known Symbol Padding,” IEEE Trans.Signal Process.,Vol.54,No.3,Pp.979-990,Mar.2006.

[2] D.K.Kim and P.Park, ”Adaptive Self-Orthogonalizing Per Tone Decision Feedback Equalizer For Single Carrier Modulation,’’ IEEE Signal Process Let.Vol.13,Pp.21-24,Jan.2006

[3] X.Gao,B.Jiang,X.You,Z.Pan,Y.Xue, ”Efficient Channel Estimation For MIMO Single- Carrier Block Transmission With Dual Cyclic Timeslot Structure”, IEEE Trans.Commun.,Vol.55,No.11,Pp.2210-2223,Nov.2007.

[4] J.Hua,X.Zhao,Z.Xu,And L.Meng, ”An Adaptive Doppler Shift Estimator In Mobile Communication Systems” .IEEE Antenna Wireless Propag.Lett.,Vol.6,Pp.117-121,2007.

[5] J.Hua,Z.Xu And J.Li. ”Doppler Shift Estimator with MMSE Parameter Optimization For Low SNR Environment In Wireless Communication,” IEEE Trans.Aerosp.Electron.Syst., Vol.44, No.3, Pp.1228-1223, July 2008

[6] J.Hua,L.Meng,X.Zhao,G.Li,D.Wang,And X.You,” A Doppler Shift Estimator In Radio Propagation,” Radio Science,Vol.44,No.RS4005,Pp.19,2009

[7] M.L.Ku,W.C.Chen,And C.C Huang,” EM Based Iterative Receivers For OFDM And BICM/OFDM Systems In Doubly Selective Channels,” IEEE Trans.Wireless Commu.,Vol10,No.5,Pp.1405-1415,May 2011

[8] J.H.Lee,J.C.Han,And S.C.Kim,” Joint Carrier Frequency Synchronization And Channel Estimation For OFDM”,IEEE Trans.Wireless Communi.,Vol.7,No.8,Pp.3196-3205,Jan2006

[9] T.Wang,C.Li,And H.H Chen, ”An Iterative Expectation Maximization Algorithm Based Joint Estimation Apporoach For CDMA/OFDM”,IEEE Trans.Wireless Commun.Vol.10,No.5, Pp167-172,Aug 2008

[10] I.Krikidis,J.L Danger,And L.Naviner, ”An Iterative Reconfigurability Apporoach For WCDMA High Data Rate Communication,” IEEE Trans Wireless Commu.,Vol.13,Pp.8-14,June 2006

[11] Tianqi Wang,Cheng Li, ”EM-Based Adaptive Frequency Domain Estimation Of Doppler Shifts With Crlb Analysis For CDMA Systems”, IEEE Trans,Vol.60.N0.1,Jan 2012

[12] Ronald A. Iltis,”A Sequential Monte Carlo Filter for Joint Linear/Nonlinear State

Estimation with Application to DS-CDMA”, IEEE Transactions on Signal Processing, Vol. 51, No. 2, February 2003

[13] Ohno and iwao sasase (2010) “Adaptive Transmission Power Control for MIMO Diversity Employing Polarization Diversity in OFDM Radio Access” department of information and computer science,kieo university JAPAN.

[14]D. Tse and P. Vishwanath. Fundamentals of Wireless Communications. Cambridge University Press, 2005.

[15] P. Viswanath and D. Tse. Sum Capacity of The Vector Gaussian Broadcast Channel and Uplink-downlink Duality. IEEE Transactions on Information Theory, 49(8):1912 – 1921, aug. 2003.

[16] M. Vutukuru, H. Balakrishnan, and K. Jamieson. Cross-Layer Wireless Bit Rate Adaptation. In ACM SIGCOMM, 2009.

[17] S. H. Y. Wong, H. Yang, S. Lu, and V. Bharghavan. Robust Rate Adaptation for 802.11 Wireless Networks. In ACM MobiCom, 2006.

[18] H. Yu, L. Zhong, A. Sabharwal, and D. Kao. Beamforming on Mobile Devices: a First Study. In ACM MobiCom, 2011.

[19] Y. Zhang, C. Ji, Y. Liu, W. Malik, D. O’Brien, and D. Edwards. A Low Complexity User Scheduling Algorithm for Uplink Multiuser MIMO Systems. IEEE Transactions on Wireless Communications, 7(7):2486 –2491, july 2008.

[20]Andrea Goldsmith “Wireless Communications”

[21]Theodere S.Rappaport “Wireless Communications” Principles and Practices Second edition

[22] http://www.cambridge.org/9780521837163